**International Symposium of Consumer Electronics 2007**

**Program**

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<td>09:30 AM-10:30 AM</td>
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<td>Enabling Technology 1</td>
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<td>09:30 AM-10:30 AM</td>
<td>Gaming Platforms</td>
<td>Security &amp; Digital Rights Management</td>
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Wednesday, Jun 20

8:30 AM - 9:30 AM

Keynote 1⭐

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Introducing Change: A Zenful Digital Experience
Room: Trinity 1-4  Chair: Sam Broyles (Texas Instruments, USA)

9:30 AM - 10:30 AM

Audio Video Technology 1 ★
Room: Trinity 6-8

Enhancing SI in Cables: DVI to HDMI and Class-B Differential Signaling ★
Raj Nair (Senior Member, IEEE / ComLSI, Inc., USA)
This paper discusses aspects impacting signal integrity of binary signals transmitted over Cat-5e equivalent cables as well as techniques to enhance the quality of signals transmitted. We inspect signal preconditioning through pre-emphasis/de-emphasis, and discuss their benefits and application. Spectral equalization through resonant filters and inter-symbol-interference cancellation through FIR filters is also reviewed. We also look at DVI and HDMI specifications, and particularly, progressive improvement in the electrical specifications within, and identify aspects in the HDMI specification that improve upon DVI as it relates to high-speed signal transmission over long cables. We also introduce symmetric true-differential signaling, called Class-B Differential Signaling (CBDS), conceived, developed and implemented in 180nm CMOS by ComLSI®, advancing the state of the art in differential signal transmission, enabling multi-Gb/s audio/video data throughput over very long cables.

A Low Cost Tile-based 3D Graphics Full Pipeline with Real-time Performance Monitoring Support for OpenGL ES in Consumer Electronics ★
Ruei-Ting Gu (National Sun Yat-Sen University, Taiwan)
This paper presents a 3D graphics engine which is specifically designed to minimize the hardware cost while providing sufficient computing capability for consumer electronics with small to medium screen sizes (up to 800x600) such as digital television. The presented 3D engine consists of a full 3D graphics pipeline for both geometry and rendering operation. This engine provides a standard AHB interface that makes it easily to be integrated into an AMBA-based SoC. The development of the 3D engine has gone through a rigorous design process: starting from system modeling (using SystemC), RTL implementation, hardware/software co-simulation and FPGA verification to test chip fabrication. This 3D engine provides 3.3M triangles/s and 664M pixels/s in maximum performance at 138 MHz using 0.18 silicon technology with 987K gates that is sufficient for most applications for digital television. At the same time, a complete OpenGL-ES 1.1 API, windowing system, Linux operating system, device driver and a 3D performance monitoring tool have been developed for our 3D engine. This performance monitoring tool provides run-time performance information include frame rate, triangle rate, pixel rate, involved OpenGL function list, function counts, memory utilization and etc. Moreover, a built-in real-time AHB bus tracer is also provided to monitor the bus activities of the 3D engine and other components on the system bus. The bus tracer captures on-chip bus signals at ether cycle accurate or transaction levels and applies real-time compression to both levels of signals. With the performance monitoring tool and the bus tracer, the 3D application developer can easily analyze the communication of the components and fine tune the 3D application to optimize the entire SoC system performance and to satisfy performance/cost constrains on consumer electronics. Both of the hardware and software have been carefully verified and demonstrated on FPGA using ARM versatile SoC develop board. The test chip will be ready for tape out in April 2007.

Architecture Design and VLSI Implementation of the Re-configurable 2-D CAT/ICAT Chip for the Applications of Image Compression ★
Rong-Jian Chen (National United University, Taiwan); Jui-Lin Lai (National United University, Taiwan)
The goal of this paper is to present the architecture design and VLSI implementation of the re-configurable 2-D CAT/ICAT chip for developing the CAT wavelets-based image coding system. The reasons that we develop the CAT wavelets-based image coding system are: (1) Extensive research has shown that the images obtained with wavelet-based methods yield very good visual quality, many researchers now believe that encoders that use wavelets are superior to those that use DCT or fractals. (2) In some systems with progressive image transmission, like WWW browsers, the quality of the displayed images follows the sequence: (a) weird abstract art; (b) you begin to believe that it is an image of something; (c) CGA-like quality; (d) loss-less recovery. With very fast links the transition from (a) to (d) can be so fast that you will never notice. With slow links the time from one stage to the next grows exponentially, and it may take hours to download a large image. Considering that it may be possible to recover an excellent-quality image using 10-20 times less bits, it is easy to see the inefficiency. Furthermore, the mentioned systems are not efficient even for loss-less transmission. The problem is that such widely used
schemes employ a very primitive progressive image transmission. On the other extreme, the CAT wavelets-based image coding system is a state-of-the-art method that was designed for optimal progressive transmission. (3) The transmitted bit stream or compressed image file is completely embedded, so that a single file for an image at a given code rate can be truncated at various points and decoded to give a series of reconstructed images at lower rates. The applications of the CAT wavelets-based image coding system can be (1) Consumer applications such as multimedia devices (e.g., digital cameras, personal digital assistants, 3G mobile phones, color facsimile, printers, scanners etc), (2) Client/server communication (e.g., the internet, Image database, Video streaming, video server, etc.), (3) Military/surveillance (e.g., HD satellite images, Motion detection, network distribution and storage, etc.), (4) Medical imagery, (5) Storage of motion sequences (e.g., digital cinema, HD digital Camcorder), and (6) Remote sensing, digital libraries/archives, and E-commerce. Due to 2-D CAT/ICAT chip is one of major key components in the CAT wavelets-based image coding system, we therefore focus our attention to the 2-D CAT/ICAT chip. To facilitate the development of the re-configurable 2-D CAT/ICAT chip, we firstly develop the evolution of CA to produce orthogonal 1-D CA bases; then we use the canonical product circuit to produce 2-D CA bases; finally, it utilizes input data and 2-D CA bases to produce 2-D CAT/ICAT coefficients. In order to enhance the flexibility of image applications, it makes the proposed chip to produce 2-D 4-by-4, 8-by-8, and 16-by-16 CAT/ICAT coefficients. Consequently, we developed the re-configurable 2-D CAT/ICAT circuit. Throughputs of the 2-D 8-by-8 and 16-by-16 CAT/ICAT coefficients are same as that of 2-D 4-by-4 CAT/ICAT coefficients due to the proposed re-configurable 2-D CAT/ICAT chip has the property of highly parallel processing. We have completed the circuit synthesis using the SYNOPSYS tolls with the UMC 0.18um Cell-library. The chip size was 12.888 mm2, and the maximum operation frequency was 111MHz with 851mW total dynamic power. It shows that the architecture of the proposed re-configurable 2-D CAT/ICAT chip is suitable for VLSI realization.

Rong-Jian Chen (M2002, SM2004 IEEE) received the B.S, M.S., and Ph.D. degrees in electronic engineering from the National Taiwan University of Science and Technology, Taipei, Taiwan, in 1987, 1991, and 1995, respectively. He joined the faculty of the National Lien-Ho Institute of Technology in August 1995, where he was the Chair of the Department of Electronic Engineering during the 1999-2001 academic years. At present, he is Associate Professor of the Department of Electronic Engineering, National United University. His research interests include digital image/video signal processing, image/video coding and encryption, pattern recognition, neural networks, VLSI architecture design and implementation for image/video coding and encryption, VLSI architecture design and implementation for neural network systems, and globalization leadership. He is a member of Nano-electronics and Giga-scale System Technical Committee, IEEE CAS Society. He serves as Co-Editor of Globalization Leadership Column at IEEE Circuits and Devices Magazine (per invitation from EIC Dr. Ron Waynant), and serves on the Editorial Board of IEEE Circuits and Systems Magazine (per invitation from EIC Prof. Maciej Ogorzalek). He also serves as Associate Editor of IEEE Trans. on Circuits and Systems, Part 1 from Dec. 15, 2005. He also serves on the Technical Program Committee of 2004 IEEE ICECS (International Conference on Electronics, Circuits and Systems) at Tel-Aviv, Israel in Dec. 2004. He served on the Organizing Committee of 2003 IEEE International Symposium on Nanoelectronics and Gigascale Systems, Miaoli, Taiwan, Oct. 2003, which was technically co-sponsored by IEEE Circuits and Systems Society and IEEE Electron Devices Society. Prof. Chen also serves on the Organizing Committee of 2004 International Workshop on Efficient and Competitive Professors to be held in Hsin-Chu, Taiwan, Dec. 2004. He serves as Review Committee Member for 2005 IEEE ISCAS Symposium, in Kobe, Japan, Review Committee Member for 2006 IEEE ISCAS Symposium, in Kos Island, Greece, and the Technical Program Committee of 2006 IEEE ICECS at Nice, French in Dec. 2006. He also serves as the Technical Program Committee of 2007 IEEE ICECS.

An Image-Based Rendering System for 3D-TV Applications Employing Commodity Computer Hardware

Christian Weigel (Technische Universität Ilmenau, Germany); Peter Schuebel (Technische Universität Ilmenau, Germany)

In this paper, we present a software system suitable for the implementation of different stages of production pipeline, namely the acquisition, processing and rendering of 3D video objects based on image-based rendering. Furthermore, we show an exemplary implementation of a view synthesis algorithm based on the trifocal transfer that employs the GPU for image warping. Due to its modular structure the system can easily be extended to the needs of each stage of the production pipeline. New modules can be added and created easily by implementing a simple common API provided by the system. The modules are attached in a graph performing a specific task. Since the production pipeline often requires processing at interactive rates, a number of enhancing tools are provided by the system. With respect to the newly introduced multi core CPUs the system supports multithreading. A single module or a group of modules can be associated with its own thread, respectively. Synchronization features and buffering possibilities make the processing of the graph highly customizable. The time critical traversal of the graph is handled by a so called real-time engine. The engine is responsible for the observation of time constraints given by the user. In addition to the newly emerging CPU technologies, GPU technologies have developed even stronger. We incorporate this fact by including an OpenGL engine into the systems. With this
Enabling Technology 1

Room: Trinity 4

Object-Based A/V Application Systems: IAVAS I3D Status and Overview

Ulrich Reiter (Technische Universitaet Ilmenau, Germany); Uwe Kuehhirt (Fraunhofer IDMT, Germany)

This paper reports on the current status of the IAVAS I3D MPEG-4 player originally developed at the Institute of Media Technology (IMT) at Technische Universitaet Ilmenau, Germany, and now jointly advanced by IMT and Fraunhofer IDMT, also located in Ilmenau, Germany. The features of the player that are of most interest to the field of consumer electronic devices, implemented in the audio and visual rendering engines, are described in detail. Also, one section is dedicated to problems and solutions related to the authoring processes of object-based interactive audiovisual content. We conclude this paper with an outlook on upcoming features.

Uwe Kuehhirt studied Electrical Engineering at Technical University Ilmenau (Germany), from where he received a Diplom-Ingenieur degree (equ. M.Sc.) in 1997. He has been working at the Institute of Media Technology (director Prof. Dr. Karlheinz Brandenburg) at the same University as a researcher in the field of interactive audiovisual applications using the MPEG-4 standard since then. He has been giving lectures on Interactive Media and Multimedia Tools at TU Ilmenau. His Ph.D. thesis is titled "Authoring of object-based AV applications". In 2005 he joined the Fraunhofer Institute for Digital Media Technology IDMT in Ilmenau and took over the management of the "Authoring Systems" group.

User Interface to Input Three-Dimensional Information with User's Palms

Morigasa Matsuda (Mitsubishi Electric Microcomputer Application Software Co. Ltd., Japan); Takako Nonaka (Ryukoku University, Japan); Tomohiro Hase (Ryukoku University, Japan)

This paper describes a contact-free input method without using a pointing device. The proposed device has an input function to capture the user's palm motions and an output function to display the screen reflected by the palm motions. First, the miniature camera captures the palm motions and inputs the images. Next, color conversion appropriate for palms, and wrinkle detection by thinning and detection of optical flows are executed. Because the previous proposals only detected the optical flows of right-to-left and up-and-down movements of the palm, the system could only move a cursor displayed on the output screen. This paper extends the system to detect palm motions toward and away from the camera. This allows new functions, e.g., selecting menus, clicking icons, and zooming the viewing area in and out, based on the three-dimensional motions of the palm. To verify the proposed user interface, evaluation experiments were conducted. First, a palm-size verification system was constructed by using an embedded processor for consumer use. Next, the optical flows of palm wrinkles were detected based on the system. Thirdly, the direction and the distance of the palm motions were estimated from the detected optical flows and were related to the screen image of the output display. The usability testing showed that the optical flows of the palm wrinkles were detected normally. Therefore, the proposal has realized a new user interface to three-dimensional input without using existing pointing devices.

Takako Nonaka was born in 1975 in Osaka, Japan. She received her BS, MS and Ph.D. degrees from Kyoto Institute of Technology, Japan in 1998, 2000 and 2004. She engaged in research on modelling the sensory evaluation as a visiting researcher of Tomas Bata University in Zlin, Czech from April to September 2004. She was a postdoctoral research fellow of High Technology Research Center at Ryukoku University from October 2004 to March 2006. Since May 2006 she has been engaged in research into user interfaces with the Research Center for Information Communication Systems of Ryukoku University as a postdoctoral research fellow.

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**A Mobile Interface for Hierarchical Information Visualization and Navigation**

Jie Hao (University of Texas at Dallas, USA); Kang Zhang (University of Texas at Dallas, USA); Te-Cheng Hsieh (University of Texas at Dallas, USA)

ABSTRACT There is a dramatic increase in the population who use mobile computing devices, such as personal digital assistant (PDA). Though the hardware is becoming more powerful, effective support for information rendering on small screens very much lags behind. To display hierarchical information, researchers have proposed many tree visualization algorithms for desktop computers. Such algorithms can be divided into two basic approaches: connection and enclosure. The connection approach can display a tree with a clear structure but consume display area, enclosure can utilize the screen space but it's hard to understand the tree's structure from the layout. This paper describes a new tree drawing technique for visualizing hierarchical data on portable devices. Our technique aims at combining the advantages of both connection and enclosure approaches in efficient use of the display space and clear layout of the tree structure. Instead of displaying the tree as a set of nodes connected by links, we divide the given rectangular display space into blocks to express nodes, connected nodes represented by adjacent blocks. This way, we keep the original structure of the tree while minimizing the space usage. Our tree visualization method uses depth first search to recursively partition the display area according to the tree structure. This method can be adapted to handle different shaped tree structures. We demonstrate our approach by applying it music classification on the mobile device at the end of this paper.


**An Emotion Expression System for the Emotional Robot**

CheonShu Park (Electronics and Telecommunications Research Institute(ETRI), Korea)

In this paper, we present an emotion expression system for expressing emotion for the emotional robot, which have a plurality of different sensors sense information about internal/external stimuli. In detail, we propose method for processing information that is collected through various sensors in an intelligent robot, a method for determining an emotion, and a method for expressing a corresponding action. Our system can express a specific emotion of the emotional robot in the form recognizable by a user so that an emotional communication between the user and the emotional robot through affective communication. An emotional expression system is comprised of an emotion feature information collector component, an internal status management component, an action determiner component and an action expression component. Our system can be applied to the emotional robot.

Also, our system can be used in developing the cyber character having an emotion or developing the apparatus having an emotion in the ubiquitous environment.

**General Topics**

Room: Trinity 1-3

**Scalable software architecture for high performance video codec on parallel processing engines**

Mihir Mody (Texas Instruments Ltd, India); Krishnakanth Rapaka (Texas Instruments Ltd, India); Keshava Prasad (Indian Institute of Technology Kanpur, India)

Video algorithm (e.g. H.264, MPEG2/4 etc) requires tremendous amount of computation power and data bandwidth. This complexity depends on encoding vs. decoding mode, video standard, resolution, frame-rate and visual quality constraints. The typical solutions for such high performance results in multiple processing elements e.g. multiple DSP or MCU, DSP/MCU with dedicated accelerator or FPGA etc. This gives new challenge to typical video software's that are designed with single processing element in mind. This paper presents software design architecture for such parallel processing elements. This paper explains following aspects in details a) Software partitioning b) Algorithm specific optimizations c) Processor specific optimizations d) Efficient DMA/Cache usage e) Concurrent scheduling of all parallel processing elements The given approach is explained with example of MPEG4 encoder on TMS320DM6446, which is DavinciTM family device from Texas Instruments Ltd. The given software architecture is scalable for various video standards (e.g. H.264, MPEG2/4 etc) as well as various parallel processing hardware solutions. The software achieves performance D1@30fps on given device at less than 50% of DSP load.

Mihir completed master of engineering in electrical engineering department from Indian Institute of Science, Bangalore, in January 2000. He joined Sasken communication, where he worked on various audio technologies like MP3, AAC etc. Later He joined Texas Instruments, India. He worked on various audio , video and sytem projects in multimedia domain. Currently, He is member of techincal staff in Multimedia Codec group focusing on video codecs. (Email Contact: Mihir@ti.com).
**Gesture Recognition Using a Smart Camera For HCI Applications**

Timothy Tsui (NICTA, Australia)

In the last couple of decades, advances in computational power and physical size reduction in CMOS image sensors and semi-conductor chips have made it increasingly viable for computer vision systems to be built into self-contained embedded units and used in a vast variety of applications such as robotics, optical character recognition, medical imaging, gaming (as recently seen in the Sony EyeToy) and Human Computer Interaction (HCI). This system, called a smart camera serves to perform image capture, image processing and output an abstract understanding of the imaged scene. In this paper, we focus on the design and implementation of a smart camera called GestureCam that can detect and recognize simple head and hand gestures for the application of HCI. In particular, the GestureBrowser, an extension to the Mozilla Firefox browser is envisioned to work with GestureCam in tandem to control web navigation. GestureCam is entirely built on an FPGA platform which we believe offers unparalleled diversity and performance in developing image processing algorithms. Current experimental results indicate that real-time performance is indeed achievable making it suitable for HCI. GestureCam’s hardware consists of a CMOS image sensor with Bayer pattern image output that is connected to a Xilinx Virtex II Pro FPGA which does all the image processing. These functions include colour interpolation, skin-tone detection and segmentation, average filtering, contour tracing, image moment calculation for feature extraction and finally, classification and recognition of head and hand gestures. Our latest addition to GestureCam has been the inclusion of contour tracing, image moment feature extraction and classification of moving hand gestures. For contour tracing, the Inner Boundary Tracing algorithm was shown to perform reasonably well, yielding close to real-time performance and robust edge detection. The algorithm relies on a full image frame in order to complete tracing and that everything but the object of interest has been filtered out. In contrast to normal gesture recognition, our classification is based on the trajectory of a moving hand rather than the orientation of the fingers with respect to the hand such as those seen in sign language. The image moment extracted from the contour provides the basis for classifying simple hand gestures such as up, down, left, right as well as drawing an arrow in the same directions. The image moments are grouped in pairs called tuples and additional features are obtained such as the x-component, y-component and gradient which form inputs to a hidden layer neural network for classification. The neural network is a non-trainable three layer network with hard-coded adjustable weights. The preliminary results show that real-time constraints can be met with a good recognition success rate. The final paper submission will include results.

**10:45 AM - 12:00 PM**

**A/V RF & Wireless 1**

Room: Trinity 1-3

**Performance Analysis and Optimization of the Wireless USB Protocol**

Yaser Ibrahim (Texas Instruments, USA)

The Wireless Universal Serial Bus (WUSB) protocol is a new wireless standard for Wireless Personal Area Networks (WPANs) targeting computer peripherals and consumer electronics. It is built on the ubiquitous USB standard and extends it to the wireless realm by utilizing Ultra-wideband technology based on the WiMedia UWB standards. The throughput performance of the WUSB link (pipe) for given channel conditions depends on the selection of the protocol parameters, namely the packet size and the burst size, as well as the raw data rate. For an error-free channel, increasing these parameters will increase the protocol efficiency and throughput. However, as the bit error rate increases this may no longer be valid, and increasing some of these parameters may not yield any benefit, and may even degrade the performance. In this paper we analyze the throughput performance and efficiency of the WUSB protocol versus the protocol parameters under different conditions in terms of bit error rates and available bandwidth. This analysis is used to optimize the throughput achieved for a certain WUSB data pipe under some given conditions.

**A Fully Hardware-oriented Medium Access Slot Management for Wimedia MAC**

Sangjae Lee (Electronics and Telecommunications Research Institute, Korea); Young-Ae Jeon (ETRI, Korea); Sangsung Choi (Electronics and Telecommunications Research Institute(ETRI), Korea)

In these days, UWB technologies for WPAN(wireless personal area network) guarantees 480 Mbps and its goal is to offer 1 Gbps. In order to support such high speed transmission, it is required to maintain real time characteristics and also required to implement high speed logic in a MAC layer. In order to overcome performance problems, more MAC functions are migrated from software to hardware and accelerate data processing between MAC software and physical layers. Time slot management is one of the timing critical functions and usually
implemented in a software level – by firmware using system timing interrupt. In this paper, we propose medium access slot (MAS) management method which is based on MBOA/Wimedia MAC. The proposed MAS management scheme is fully implemented in hardware and is independent of software methodologies that use timing interrupt. Furthermore, it can be implemented by using only a small amount of logic. Based on Wimedia MAC specification, all the reservation information for time slot assignment in one device is included in DRP (distributed reservation protocol) IE (information element) in a transmit beacon. In order to assign reservation information of time slot in a real time manner, TX beacon parsing method is used in our proposed method. We use DRP IEs contained in a TX beacon payload for analyzing reservation information. By sequentially fetching TX beacon frame from TX beacon FIFO to which TX beacon frame is already written by software, DRP IEs are identified by checking Element ID field. Then, MAC hardware automatically reserves MASs according to the DRP IE. After assigning dedicated queue for new stream by MAC software, whole MAS management process is performed in MAC hardware. If TX beacon is transferred from main memory to TX beacon FIFO, TX beacon parser analyzes the DRP IE fields of TX beacon and generates reservation information. Concurrently with the generation of reservation information, time slot assigner compares the reservation information with the information in assigned queue list. If matched information is found, it generates MAS reservation information that can be used by the superframe controller and sets the MAS information list in superframe controller. All MAS information of one DRP IE is updated only within one clock cycle. Moreover, time slot assigner compares the reservation information from DRP IE with the assigned queue information. If match information is found, it generates information such as DRP queue BD address and successive MAS counters. After reserving all MASs for the superframe, the superframe controller manages the whole TX/RX time to operate MAC hardware. The superframe controller generates control signals for reading TX beacon frame and data frames according to MAS information. It performs micro-second scheduling for superframe and MAS and zone scheduling for enabling hardware-oriented MAS management. In Wimedia MAC, one superframe consists of 16 zones and each zone again consists of 16 MASs. Therefore, totally 256 MASs exist in one superframe and each MAS information is matched into one real MAS. The time slot assigner updates each MAS according to zone and MAS bitmap transferred from TX beacon parser. Each MAS information contains MAS Valid Tag, Reservation Type, Target/Owner bit. In addition, DRP Queue Address represents the TX data BD address, and Successive MAS Counter is used for calculating the remaining time. Finally, we implemented the Wimedia MAC and PHY board. The whole MAC hardware is implemented in Xilinx Vertex4 XC4VLX60 device. The MAC core utilizes 9,293 slices (34 %), 10,982 flip flops (20 %), and 15,396 4 input LUTs (28 %). It operates up to 66 MHz core clock frequency and interfaced with MBOA PHY module which can operate up to 480 Mbps. And the average application bandwidth is about 70 % of PHY bandwidth.

**A Practical Low Cost Architecture for a MB-OFDM Equalizer (ECMA-368)**

R. Simon Sherratt (University of Reading, United Kingdom)

Ultra-Wideband (UWB) communication has been synonymous with Wireless Area Personal Networks (WPAN) since 2002 when the Federal Communications Commission (FCC) relaxed the regulation for UWB transmission. A WPAN network offers the Consumer Electronics Industry a low-power fast and plug-and-play wireless network for Consumer Devices within a short range. After all the proposals for UWB WPAN systems, two main systems still remain those of DS-UWB and MB-OFDM. Although there are distinct advantages to DS-UWB, for WPAN industry has clearly sided with MB-OFDM. There are many reasons for this, but notably its selection for Wireless-USB, physical layer for the new fast Bluetooth, WiMedia consortium has grown considerably and many pre-existing OFDM designs (DVBT/H, DAB, ISDB-T and certain 802.11 variants) have notable relevance to MB-OFDM. Using a 128-pt FFT, the MB-OFDM sub-carrier rate of 528MHz is considerable even for today’s silicon. A parallel FFT scheme to allow receiver decoding (or transmitter coding) at clock rates of 528/4 MHz has been previously presented allowing for much lower clock rates, even to the point allowing prototyping in FPGA. The receiver equalizer follows the FFT decoding. Even after the removal of 6 null sub-carriers the receiver equalizer still requires a high comparable clock rate. Complicating the issue further is that to fully exploit the frequency diversity inside each OFDM frame, three different equaliser operations are needed. To improve performance, one may also include the guard sub-carriers to aid symbol reliability. This paper will present our selected equalizer architecture for efficient equalization of MB-OFDM received signal for use with all of the current coding schemes. We will present the performance of the equalizer to relevant IEEE 802.11.3a propagation channels as a function of available numerical precision. The selected architecture will be described in both textual and diagrammatical form. We conclude that we need 9 bit integer and 5 bit fractional numerical performance in the receiver FFT and 12 bit integer performance in the equalizer to maintain the same performance as a similar floating point model.

References will be supplied in the paper


http://edas.info/showProgram.php?of_showProgram=&c=5136&foroma...bio%5D=1&program_view=everybody&action=Show+conference+program
A Method of Detecting Time-Frequency Code in MB-OFDM UWB System
Sung Woo Choi (ETRI, Korea); Sangsung Choi (Electronics and Telecommunications Research Institute (ETRI), Korea)

The Multi-Band OFDM UWB uses 3.1-10.6 GHz frequency as 14 sub-bands that have a spacing of 528 MHz. In each sub-band, information is transmitted using OFDM modulation. The fourteen sub-bands are organized into five band groups: four groups of three sub-bands and one group of two sub-bands. Band group 1 is used for mandatory mode and the remaining band groups are reserved for future use. Using the same band group, multiple piconets should be made in some area. The piconet means a network of devices connected in an ad hoc fashion. Channelization for different piconets is achieved by using different time-frequency codes for different piconets. In addition, different preamble patterns are used for the different piconets. Time-frequency code means that we can use different sub-band at every OFDM symbol. There are 7 time-frequency code in band group 1. Using this time-frequency code, we can detect the present used channel and communicate with other systems in that piconet. So, when a new device wants to enter a present piconet, it should detect the time-frequency code of that piconet. This detecting process usually managed by MAC layer. The MAC controls modem and RF, according to time-frequency code, to recover information of the piconet. Because the total number of channel is 7, the channel scanning process is composed of 7 stages. Because the MAC and PHY exchanges data and control signals at each stage, the scanning process of 7 stages is more complicated. In this paper, we propose a new method that controls physical layer using modified time-frequency code, which makes it easy to detect the used time-frequency code. This method detects time-frequency code in 4 stages which control the preamble pattern of the modem cross-correlation block and the operation sub-band of RF. Between the four stages, we explain the stage one briefly for an example. If a piconet used time-frequency code 1, 2 and 5 then, according to specification of MB-OFDM UWB, the preamble pattern 1, 2 and 5 should be received through the sub-band 1 at least once in time of 3 OFDM symbols. So, if we fix the sub-band of RF to band 1 and switch the preamble pattern of the cross-correlator every 4 OFDM symbols in order of preamble 1, preamble 2 and preamble 5, then we can detect peak in modem cross correlation block. The control block of stage one outputs the band selection signal of RF and the preamble pattern signal of the cross-correlator as following table. 

<table>
<thead>
<tr>
<th>Symbol time 1</th>
<th>2</th>
<th>3</th>
<th>4</th>
<th>5</th>
<th>6</th>
<th>7</th>
<th>8</th>
<th>9</th>
<th>10</th>
<th>11</th>
<th>12</th>
<th>13</th>
<th>...</th>
</tr>
</thead>
<tbody>
<tr>
<td>RF band</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>Preamble</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>2</td>
<td>2</td>
<td>2</td>
<td>2</td>
<td>2</td>
<td>5</td>
<td>5</td>
<td>5</td>
<td>5</td>
<td>5</td>
</tr>
</tbody>
</table>

Repeat from selection symbol time 1
This method detects time-frequency code in 4 stages which control the preamble pattern of the modem cross-correlation block and the operation sub-band of RF. Between the four stages, we explain the stage one briefly for an example. If a piconet used time-frequency code 1, 2 and 5 then, according to specification of MB-OFDM UWB, the preamble pattern 1, 2 and 5 should be received through the sub-band 1 at least once in time of 3 OFDM symbols. So, if we fix the sub-band of RF to band 1 and switch the preamble pattern of the cross-correlator every 4 OFDM symbols in order of preamble 1, preamble 2 and preamble 5, then we can detect peak in modem cross correlation block. The control block of stage one outputs the band selection signal of RF and the preamble pattern signal of the cross-correlator as following table.

The implementation of scalable RF terminal system based on 3G Evolution standards
JaeHo Jung (Electronic and Telecommunications Research Institute, Korea)

In this paper, the results on the implementation of scalable RF terminal system based on 3G Evolution standards will be presented. The developed RF terminal has wideband multiple transmit and receive paths up to 40MHz to support MIMO and bandwidth scalability. Therefore, the implemented RF system has the characteristics of structure to apply the dual duplexing system of TDD and FDD at RF front-end and this system has the digital IF structure similar with the base-station to support the scalable bandwidth using the multiple frequency allocations. We will propose the measurement methods of capacity of the implemented RF system through accurate EVM measurement. And this measurement results is compared with the simulation results on the RF characteristic modeling of dominant factor, phase noise, IQ mismatch and noise figure, and so on. In this paper, we define the channel estimation loss to measure the accurate EVM by the RF impairments only and AWGN robust channel estimation method with smoothing and averaging is used. The implementation results show very low constellation error at maximum output power and accordance with the RF simulation results.

Audio Video Technology 2

Room: Trinity 6-8
Chair: Jianping Zhou (Texas Instruments, USA)

A fast mode decision algorithm using a block correlation in h.264/AVC
Jong-Ho Kim (ETRI, Korea); Byung-Gyu Kim (Electronics and Telecommunications Research Institute, Korea); ChangSik Cho (Real-time Multimedia Lab, Electronics and Telecommunications Research Institute (ETRI), Korea)

MPEG-4 Part-10 H.264/AVC is the latest video coding standard provides a higher coding efficiency than previous standards. H.264/AVC achieves a bit rate saving of more than 50% with no quality loss using a rate-distortion process, but it is computationally complex. We proposed an algorithm that can reduce the complexity of the codec by adaptively reducing the block mode decision process. An efficient method is presented to reduce the coding complexity using direct prediction methods based on inter block correlations and an adaptive rate distortion cost...
Development of H.264 Encoder for a DSP Based Embedded System ★

Nicholas Vun (Nanyang Technological University, Singapore); T N Anh Nguyen (Nanyang Technological University, Singapore)

Compare to other earlier H.26x series and MPEG standards, H.264/AVC introduces many enhancements and new tools to provide superior performance in terms of compression efficiency and error recovery capability. The main feature of H.264 is its flexibility to operate at different modes based on the requirements. This is done through the different 'profiles' defined that primarily allow the operating bitrates and resolution to be varied. This is particularly well suit for embedded system implementation such that only the relevant features are needed to be implemented into the system. One of the most promising applications for H.264 on embedded system is the streaming of compressed digital video over network to backend system for further processing. However, by utilizing digital signal processor in the embedded system, the signal processing can be performed locally in the embedded system prior to streaming, which can further reduce the amount of data needs to be streamed over the network. This paper describes the process of developing a H.264 Baseline profile encoder on the TMS320DM642 DSP platform. Due to the complexity of the H.264 encoder, it is decided that instead of developing the code from scratch that specifically targeted for a DSP family, a Win32 reference encoder software is used as the starting point of the development. This papers will emphasize on the various issues and problems encountered during the porting process, as well as the optimizations techniques used to vastly improve the performance of the embedded system. Final results of encoding 10 video frames are as follows: a) 26 seconds on the Win32 reference encoder running on a 2.5GHz 512MB P4 desktop. b) 37 minutes 10 seconds on a 600 MHz 36MB TMS320DM642 DSP platform - after successful porting, but without optimization c) 1 minute 5 seconds on the same 600 MHz 36MB TMS320DM642 platform, after optimization.

System Architecture Design Methodology for H.264/AVC Encoder ★

Samuel Chang (National Taiwan University, Taiwan); Chih-Chi Cheng (National Taiwan University, Taiwan); Liang-Gee Chen (National Taiwan University, Taiwan)

MPEG-4 H.264/AVC[1] has become a joint standard for ITU-T and MPEG and has been implemented in a wide spectrum of applications, from digital video broadcasting for handsets (DVB-H) to Hi-Definition DVD Storage (HD DVD). However, the resolution and system clock rate requirements vary diversely. Therefore, developing an optimal system architecture for all applications is impossible. Among all aspects of the H.264 encoding system architecture design, inter-prediction occupies 99% total computation complexity and the search-range buffer occupies 76% of total on-chip memory[1][3]. Our design methodology fully explores the design spaces of parallelism of inter-prediction, macro-block (MB) pipelining, and search-range buffer architecture. This paper combined with [4] presents a complete methodology to help H.264/AVC systems designers obtain the most area efficient design under user defined timing requirements. First, we explore the implementation and tradeoffs of inter-prediction parallelism. Inter-prediction is used to search for the motion vector (MV), which is the spatial shift between a 16x16 MB in the current frame and the corresponding MB in the previous frame. Inter-prediction consists of integer motion estimation (IME) and fractional motion estimation (FME); IME searches for motion vector with integer pixel precision, and FME refines the result of IME with quarter pixel precision. IME parallelism [3] is done by expanding the number of sum of absolute difference (SAD) adder trees at the cost of additional registers and SRAM bandwidth. The sub-pixels demanded for FME processing units (PU) are provided by the interpolation unit, which decomposes the 2-D FIR into the horizontal and vertical 1-D FIR, with an interpolation buffer [2].

http://edas.info/showProgram.php?_of__showProgram=&c=5136&form...bio%5D=1&program_view=everybody&action=Show+conference+program

Fast Inter-Mode Decision Algorithm For P Slices In H.264/AVC Video Standard ✍️

Byung-Gyu Kim (Electronics and Telecommunications Research Institute, Korea); ChangSik Cho (Real-time Multimedia Lab, Electronics and Telecommunications Research Institute (ETRI), Korea)

In this paper, a fast inter-mode determination algorithm based on the macro-block (MB) tracking scheme and rate-distortion (RD) cost is proposed for H.264/AVC video standard in which the residual prediction part is composed of intra-modes and inter-modes. In addition to intra mode prediction, 8 block types exist for the best coding gain based on rate-distortion (RD) optimization in inter mode prediction. This scheme gives rise to exhaustive computations (search) in the coding procedure. To reduce this computational load of the inter mode search at the inter-frame, we propose a new inter mode determination algorithm on the basis of the rate-distortion (RD) cost of the neighborhood MB which is tracked for the current MB in the previous frame. Base on the MB tracking scheme, an efficient sequential mode search approach is presented here. We verify the performance of the proposed scheme through comparative analysis of experimental results using JM reference software. To verify the performance of the proposed fast mode decision algorithm, various MPEG standard sequences were used with CIF and QCIF sizes in the same condition. JM 11.0 reference software by JVT (joint video team) was used as a reference code for evaluation of the encoding performance. We have used two methods for an objective comparison of the encoding performance. These are Jing’s (Alg-1) and Salgado’s (Alg-2) methods which are well-known as efficient and fast algorithms. The results show the performance of all algorithms for the IPPP sequence type. The proposed fast inter mode decision algorithm yielded more credible performance because of an adaptive RD thresholding scheme using the RD cost of the most correlated MB which was tracked in the previous frame for the current MB except SKIP mode. Also, technique for the selection of the partial candidate modes could provide an improved speed of the motion estimation procedure. From this result, we can see that the proposed scheme can achieve a significant improvement of up to 70.59% with the average loss of 0.052 (dB) and bit increment of 0.708%, in the total encoding time.

received his B.S. degree from Pusan National University, Korea, in 1996 and M.S. degree from Korea Advanced Institute of Science and Technology (KAIST) in 1998. In 2004, he received the Ph.D. degree in the Department of Electrical Engineering and Computer Science from Korea Advanced Institute of Science and Technology (KAIST). In March 2004, he joined in the real-time multimedia research team at the Electronics and Telecommunications Research Institute (ETRI), Korea where he is currently a senior researcher. His research interests include image segmentation for the content-based image coding, wireless multimedia sensor network, real-time multimedia communication and intelligent information system for image signal processing. He is a member of IEEE, IEICE, and KMMS (Korea Multimedia Society).

Enabling Technology 2 ✍️

Room: Trinity 4

MotoCAP STA Framework: New approach to testing of embedded systems ✍️

Konstantin Bulenkov (Motorola, Russia)
Any embedded system must be tested thoroughly. The traditional approach supposes manual tests or usage some kind of an automated test environment like JUnit. Such an approach has a drawback that in case of high complexity of a system it requires many man-months to reach acceptable test coverage and in some cases it is impossible to cover everything. This problem is especially important in case if the complete system consists of many individual nodes. To overcome the tests complexity an automated framework has been developed. The main goal which was born in mind during the framework development was flexibility. The framework can be fed by user scenarios and has an ability of being altered “on the fly”. The framework is protocol independent as well (TCP/IP, RS-232, IR etc). The framework was successfully used in many Motorola’s programs for Digital TV receivers (Set-Top Boxes) functionality testing. MotoCAP STA Framework enabled us to find errors in Motorola OCAP platform implementation, which were not found earlier with the use of manual or JUnit testing. Besides, with the use of this framework we managed to run a huge amount of performance measurement tests.

Konstantin P. Bulenkov graduated from the Saint-Petersburg State University. He has been working at Motorola as a technical leader of several projects since 2006. He has more than 8 years experience of using Java technologies in commercial development. He is an expert in both Java and Microsoft.NET platforms.

Reengineering for System Requirements Reuse: Methodology and Use-Case

Larisa Melikhova (Motorola, Russia); Albert Elcock (Motorola, USA); Andrey Dovzhikov (Motorola, Russia); Georgiy Bulatov (Motorola, Russia); Dmitry Vavilov (Motorola, Russia)

Having a precise and unambiguous set of system requirements is an important step in product development. If a product is produced in a variety of models with different sets of features, it is desirable to make the requirements reusable. This imposes specific restrictions on requirements development that are described in the paper. The principles that are to be followed are classified and illustrated by examples from real work.

Developing Ontology for Intelligent Home Service Framework

Inbong Joo (University of Science and Technology, Korea); Jongyoul Park (ETRI, Korea); Eui-Hyun Baik (ETRI, Korea)

Home services are all kind of services which are provided in home environment. Legacy Researches was focused on controlling home devices and appliances. And recent works focus on the multimedia services. In the future, devices existing in the home would infer the status and intention of users and provide intelligent home services without much intervention of users. To do this, there must be context-aware technologies [1]. This research focused on the ontology which is the core technology of the context information management [2]. Through developing ontology for the home services, we classified services into several categories and, described requirements for the service execution. And then, we adopted the ontology for the intelligent home service framework called iHSF. Ontology designed for the home services consists of the concept of users, places (semantic locations), environment around the specific places, devices and individual services and subordinate concepts of them, which all concepts have properties and relations among them [2]. Ontology for the iHSF has form of RDF (Resource Description Framework) [3, 4, 5] and OWL (Web Ontology Language) [6, 7] in order to be interpreted by devices. Ontology should mirror the constantly varying environment of the home dynamically. iHSF using well-designed ontology reacts to the constantly varying context on a real time basis.

12:30 PM - 1:30 PM

Keynote 2

HANA: Creating an HD Content "Trust Zone"

Room: Trinity 5

1:30 PM - 3:00 PM

A/V RF & Wireless 2

Room: Trinity 1-3

Baud Rate Symbol Timing Synchronization for 8-VSB ATSC DTV Receivers

Tom Wilson (University of Technology, Vienna, Austria)

A timing recovery system for ATSC standard HDTV receivers is presented. The proposed solution operates at baud rate, thereby enabling use of a symbol-spaced channel equalizer and simplifying the demodulator
Reducing Bluetooth Interference with Spatial Diversity Techniques in Wireless LANs ★

Mikael Gidlund (Royal Institute of Technology, Sweden)

Bluetooth is a low-cost and ow-power wireless connectivity technology operating in the 2.4 GHz unlicensed ISM band. On of the challenges to Bluetooth adopters is interference rejection and coexistence with other WLAN standards such as 802.11. In this paper, we examine how spatial diversity can be exploited to suppress Bluetooth interference on a IEEE 802.11b network. We will consider two dual antenna receivers, maximum ratio combining (MRC) and interference rejection combining.

Transmit antenna selection scheme for the retransmission in MCW MIMO OFDM systems ★

Bangwon Seo (ETRI, Korea); Heesoo Lee (ETRI, Korea); Hyun Kyu Chung (ETRI, Korea)

In packet communication systems, packet retransmission is often requested when a received packet is detected to be in error. The hybrid automatic retransmission request (HARQ) schemes using data from the previous (re)transmission instead of discarding them can be used to provide high reliability, or lower packet error rate and higher throughput. Recently, the multiple-input multiple-output (MIMO) multiplexing has been shown to tremendously increase the spectral efficiency of the wireless communication systems. In MIMO multiplexing, transmit data sequence is transmitted from a different transmit antenna at the same time with the same carrier frequency. Therefore, the total transmission rate increases in proportion to the number of transmit antennas without requiring additional bandwidth. Combined with HARQ, MIMO can potentially provide higher throughput packet data services with higher reliability. Until now, most of research on HARQ for MIMO OFDM systems has been done for the case that the single codeword (SCW) is transmitted using all transmit antennas. Recently, MIMO OFDM system using the multiple codeword (MCW) transmission and the successive decoding and cancellation (SDC) receiver has been considered as one of the attractive candidates for MIMO techniques in IEEE 802.20 and 3GPP LTE because it is known to achieve the total capacity (sum-rate) of the channel at any vertex of the capacity region. At the base station (BS) of this system, the multiple packets are simultaneously transmitted after each packet is encoded and QAM-mapped independently according to the channel quality information (CQI) reported from the MS every time interval T_{cqi}. (In this paper we use the post-processing SINR as the CQI.) The SINR feedback period T_{cqi} is assumed to be larger than the channel coherence time and the allowed total retransmission time. At the mobile station (MS), the SDC receiver is used. It is similar to the conventional successive interference cancellation (SIC) receiver in that the detected symbol is successively cancelled from the received signal, however, different to the conventional SIC receiver in that the SIC operation is performed after the decoding of the corresponding packet and the cyclic redundancy check. We consider the case that if a packet fails to be decoded the MS sends NACK signal for the packet to the BS and there is no additional feedback information. The conventional retransmission scheme proposed in IEEE 802.20 contribution retransmits the packets using the same transmit antennas as used at the first transmission. However, if the BS can find the transmit antenna having a better channel quality and retransmit the packets failed to be decoded using that antenna, the decoding performance of that packet can be improved with no doubt. In this paper, we propose an
improved transmit antenna selection scheme for the HARQ in MIMO OFDM system with the MCW transmission and SDC receiver. In order to find the transmit antenna having a better channel quality the recently received post-processing SINR is used. If we assume without loss of generality that the packets are decoded in the increasing order of their indices, then the post-processing SINR for the packet m, SINRₘ, is defined as the SINR obtained by applying the MMSE equalizer for packet m when the signals of the packets m, m+1, ..., M are present in the received signal. If the BS receives the NACK signal for a packet, (for example packet m), for p=m, m+1, ..., M it compares the recently received SINRₚ with SINR_q for all q satisfying q>p. If there is p and q such that SINR_q >= SINRₚ, the BS retransmits the packet p using the transmit antenna used to transmit the packet q in the previous transmission. (We define this transmit antenna index as I_q.) This is because the channel quality of the transmit antenna I_q is better than that of the transmit antenna I_p as we will prove analytically in this paper. Moreover, even though SINR_q <= SINRₚ, the channel quality of the transmit antenna I_q can be better than that of the transmit antenna I_p if the difference between two SINRs, d_SINR, is very small. Therefore, we find the optimal SINR difference d_SINR satisfying the previous condition by the simulation. We analytically find the probability such that the SINR_q >= SINRₚ for q<p and also analyze the SINR performance for the retransmitted packets of the conventional scheme and the proposed one as a function of the SNR and the number of the retransmission. In order to verify the performance of the proposed scheme, we conduct simulations for 2x2 MIMO OFDM systems transmitting two codeword simultaneously in Rayleigh fading channel. First it is shown that the analysis results for the probability and SINR coincide with the simulation results. The SINR of the proposed scheme is maximally 2dB better than that of the conventional scheme as the number of retransmission increases. When the received SNR is fixed at 0dB, the average number of retransmission for the successful decoding is 1.82 for the proposed scheme and 2.1 for the conventional scheme. We also show that the throughput of the proposed scheme is 10% better than that of the conventional scheme.

Bangwon Seo was born in Korea in 1975. He received the B.S. and M.S. degrees from the Korea Advanced Institute of Science and Technology (KAIST), Korea, in 1997 and 1999, respectively, all in electrical engineering. Since 2004, he has been a research engineer at the Electronics and Telecommunications Research Institute (ETRI) in Korea. His research interests include OFDM/Multicarrier CDMA systems, multiuser detection, and channel estimation.

**Jitter Effect on a Digital IF Transceiver for SDR-based Mobile Communication Base Station ⭐**

Bong-Guk Yu (ETRI, Korea); Gweon-Do Jo (ETRI (Electronics & Telecommunications Research Institute, Korea); Jin-Up Kim (ETRI, Korea); Sung-Woong Rha (ChungNam National University, Korea)

SDR (Software Defined Radio) technology is a promising feature that make possible to accommodate several standards through software changes on a identical hardware platform for next-generation mobile communication systems. Mobile network operators can provide state-of-the-art services to their customers rapidly with low cost of deployment and maintenance, if SDR technology is applied to their mobile communication base stations. These advantages urge many research institutes to proceed to commercialization of software radio technology worldwide. ETRI in Korea developed a double-mode base station termed a ReMo (Reconfigurable Mobile Convergence for 2-mode access system) to verify the feasibility of SDR technology to mobile communication base station. ReMo is reconfigurable to an IEEE 802.16d WiMAX (Worldwide Interoperability for Microwave Access) system based on OFDM technology, and to a HSDPA (High Speed Downlink Packet Access) system based on CDMA (Code Division Multiple Access) technology. ReMo uses SDR technologies in which modems and other functional blocks can be reconfigured easily, with software downloaded onto identical hardware platforms. This paper presents an SDR-technology-based digital IF transceiver for mobile communication base station that is reconfigurable to three bandwidth profiles: 1.75MHz, 3.5MHz, and 7MHz, each incorporating the IEEE 802.16d WiMAX standard and to HSDPA standard. An implemented transceiver manages the digital IF function in the ReMo, which incorporates heterodyne architecture. This transceiver can also be reconfigured to other mobile access standards through software downloaded onto identical hardware platforms, without changing any components or parts on board. The main functions of this transceiver include the frequency upconversion of a baseband signal from a modem to an analog IF (intermediate frequency) signal in addition to the frequency downconversion of an analog IF signal from an RF transceiver to a digital baseband signal. Here, the RF transceiver downconverts the RF signal output from a receiver-antenna to an analog IF signal. In addition, the Digital IF transceiver has a channelization function that splits one ADC (analog to digital converter) output signal into two paths for two FA (Frequency Assignment) signal processing events on the downconversion side, and combines the two FA digital signals into one combined digital signal. The one combined digital signal is then entered into the DAC on upconversion side, as the ReMo supports two FA diversity paths in both the uplink and downlink directions. In this paper, the overall Digital IF transceiver architecture and design specifications, including an implemented PCB (printed circuit board) assembly module, are presented. Furthermore, the reconfiguration function to the three WiMAX profiles and HSDPA through a ten-second software download is verified by examining the frequency response spectrum and constellation of the downlink IF output signal of the transceiver. Finally, the experimental results that show the performance of the designed Digital IF transceiver using an undersampling...
scheme, which is closely related to sampling clock jitter characteristics, are described. This paper examines the relationship between the SNR performance of SDR-based Digital IF transceiver with under-sampling scheme and the sampling jitter effect on multi-channel OFDM signal as well as multi-code CDMA signal.

**Implementation of Carrier Recovery with Large Frequency Acquisition Range for Cable Modem Downstream**

**Eungdon Lee** (Electronics and Telecommunications Research Institute, Korea)

Owing to the high spectral efficiency, high-order QAM has been applied to a variety of applications such as digital transmission over hybrid fiber coaxial (HFC) networks, digital TV system, home phoneline networking alliance (Home PNA) and many other systems. CableLabs has selected 64QAM/256QAM as the modulation format in DOCSIS (Data-Over-Cable System Interface Specification) standard for downstream delivery of video and data services through HFC networks, and the European Telecommunications Standards Institute has chosen 64QAM as the modulation scheme in the digital video broadcasting standard. One of the problems related to high-order QAM for these broadcast type services is that the carrier recovery must be often achieved blindly. There have been several blind carrier recovery methods available for high-order QAM. Sari & Moridi proposed a phase and frequency detector (PFD) algorithm using a windowing signal set to exclude non-diagonal symbols, Jablon proposed a reduced constellation (RC) PLL, which uses the decision directed (DD) phase detection and the four corner symbols to enhance robustness of acquisition, and Kim and Choi also proposed a polarity-decision phase detector algorithm, which uses several corner symbols for phase detection according to a threshold power value and exhibits more robustness and large acquisition range. Digital PLL’s such as the blind carrier recovery methods have discrete-time nature and often contain a loop delay that is produced to facilitate implementation. The presence of the delay will reduce the stability region leading to restriction of loop filter parameters of carrier recovery so that the restriction can be critical to system performance when loop bandwidth of loop filter is high during acquisition. Thus, to achieve carrier recovery with large frequency acquisition range, additional automatic frequency controller (AFC) is required in the presence of large loop delay or parallel processing can be used with high clock rate to reduce loop delay. However, both methods will increase the cost and complexity of the carrier recovery loop. In this paper, we designed a carrier recovery with small loop delay based on polarity-decision phase detector algorithm with large frequency acquisition range that is useful for cable modem downstream. To reduce loop delay we design a phase detector with look-up table, a digital controlled oscillator (DCO) with simple modular operation, and an index search of sine/cosine table. Simulations show that the newly designed carrier recovery can yield the bit error rate (BER) performance compared to the original carrier recovery in large carrier frequency offset under high-order quadrature amplitude modulation (QAM).

Senior Member of Engineering Staff in ETRI Doctor's Course in Information & Communications Engineering at Chungnam National University A Field of Study Speech Codec. Channel Coding. Cable Modem. Synchronization.

etc

**Audio Video Technology 3**

**Room: Trinity 6-7**

Chair: Jianping Zhou (Texas Instruments, USA)

**A SVC Adaptation Decision Engine Based on MPEG-21 DIA for Universal Multimedia Access**

**HaHyun Lee** (University of Science and Technology, Korea); **Jungwon Kang** (ETRI, Korea); **Jae-Gon Kim** (Hanbat National University, Korea)

This paper proposes the development of an adaptation decision engine for scalable video in universal multimedia access (UMA) environment. In UMA environment, due to the heterogeneous networks and diverse terminals, flexible and efficient video adaptation, which is performed according to network conditions and terminal capabilities as well as user preferences, is required to maximize consumer experience and ensure Quality of Service (QoS). In order to reflect this aspect, we developed an adaptation decision engine for scalable video using MPEG-21 Digital Item Adaptation (DIA) that provides systematic solutions in choosing an optimal adaptation operation among multiple options in spatial/temporal/quality domain and their combinations under given constraints and supports inter-operability of the adaptation. Currently, scalable video coding (SVC) is being developed as an extension of H.264/MPEG-4 Part10 by joint video team (JVT) between the ITU-T Video Coding Experts Group (VCEG) and the ISO/IEC Moving Picture Experts Group (MPEG). Scalable video bitstream contains embedded versions of the source content that can be decoded at different spatial resolution, frame-rate and SNR quality. By selecting the required part of the bitstream, SVC allows a very simple, fast and flexible adaptation to the heterogeneous networks and diverse terminals. MPEG-21 DIA defines a variety of tools for video adaptation. By using these tools, we can describe informations such as network characteristics, terminal capabilities and user preferences. In order to decide the optimal adaptation operations, the proposed adaptation decision engine obtains the information using AdaptionQoS, usage environment description (UED) and universal constraint description (UCD) tools defined in MPEG-21 DIA. AdaptationQoS specifies the relationship between constraints (e.g., bandwidth and display.
resolution), feasible adaptation operations (e.g. frame dropping) satisfying these constraints, and possible associated utilities (e.g. PSNR). UED contains the display resolution and available bandwidth in terminal. UCD includes the user preferences for spatial, temporal resolution, quality and item of which users want to maximize experience among spatial/temporal/quality. After receiving these informations, adaptation decision engine decides the optimal adaptation operations satisfying the given constraints. Then, these operations are inputted into SVC extractor that extracts the required part of the bitstream based on the decided adaptation operations. Especially, we proposed the adaptation decision scheme that maximizes the temporal resoultion or quality resolution for given constraints. For instance, when a user prefers high temporal resolution of video than quality for given constraints, adaptation decision engine changes the quality prior to frame rate to satisfy the given constrains. If a user prefers high quality of video than temporal resolution, adaptation engine changes the frame rate prior to quality to satisfy the given constraints. In order to evaluate the proposed adaptation decision scheme, we developed the framework for adaptation, delivery and consumption of SVC. The framework shows that proposed scheme works well under time-varying network conditions for diverse terminals and user preferences by adapting the SVC bitstream in an efficient and flexible way.

**A development of human following components for URC-based intelligent home service robots**

Hosub Yoon (ETRI, Korea)

Ministry of Information and Communication (MIC) in Korea has launched, Network-based service robot project based on the concept of Ubiquitous Robot Companion (URC). URC means that it will provide the necessary services at the server side to meet the user’s requirements. Thus, it combines the network function with the current concept of a robot in order to reduce hardware cost. The Korean IROBI produced by Yujin Robotics, a URC-based family robot has featuring education, home security, daily life management, entertainment, and message delivery. At that time, the human following component is required by customer a basic function. From this concept, we develop a human following component using combined approach as frontal face detection, omega shape detection, and head circle detection. Our algorithm should be used at the general home space such as living room or bed rooms that have the additional limits of a moving camera, moving objects, various face orientations, and unfixed illuminations. With these circumstances, if frontal faces are existed on the scene, frontal face detector based on Haar-feature and adaboosting classifier is selected one solution. Others case, one assumption that human head and body contours have an omega shape is satisfied. In order to detect omega shape, we trained a lot of upper body regions that include omega shapes using as same as frontal face detection approach. The other case, we detect head circles using three basic features of gray, color and edge thresholding method. The frontal face cue is very useful when the image stream have frontal face regions. The omega cue is also important when captured human have backward shapes. The head circle cue is helpful when captured humans have side shapes when human turn to destination. Since these three methods have roughly orthogonal failure results, they serve to complement each other. The results of our algorithm provide robustness for human rotation, illumination changing, and variable human sizes. Furthermore, it is possible to carry out real time processing.

**Multiple Initial Point Prediction based Block Motion Estimation Algorithm**

Humaira Nisar (Gwangju Institute of Science and Technology, Korea); Aamir Malik (Gwangju Institute of Science & technology, Korea); Tae-Sun Choi (GIST, Korea)

A novel, computationally efficient and robust scheme for multiple initial point prediction has been proposed in this paper. A combination of spatial and temporal predictors has been used for initial motion vector prediction, determination of magnitude and direction of motion and search pattern selection. Initially three predictors from the spatio-temporal neighbouring blocks are selected. If all these predictors point to the same quadrant then a simple search pattern based on the direction and magnitude of the final predicted motion vector is selected. However if the predictors belong to different quadrants then we start the search from multiple initial points to get a clear idea of the location of minimum point. In this case a small rood search pattern has been selected. The predictive search center is closer to the global minimum and thus decreases the effect of monotonic error surface assumption and its impact on the motion field. Its additional advantage is that it moves the search closer to the global minimum hence increases the computation speed. Further computational speed up has been obtained by considering the zero motion threshold for no motion blocks, and, specialized rood search pattern. The image quality measured in terms of PSNR also shows good results.

**A New Cryptography System and its IP Core Design for Multimedia Application**

Hun-Chen Chen (National United University, Taiwan); Jui-Cheng Yen (National United University, Taiwan); Jui-Hsiang Juan (National United University, Taiwan); Kuo-Tai Fan (National United University, Taiwan)

In this paper, we have proposed a new cryptography system which combines both the position permutation and the value transformation encryption methods. Three good features involve in this system: (1) High secruity
Structure Adaptive Up-Scaling using Principal Component Analysis

Jeroen Tegenbosch (Philips Eindhoven, The Netherlands); Paul Hofman (Philips Research Labs, The Netherlands)

I. INTRODUCTION Picture quality is still an important differentiating factor on the television market, where display resolution plays a prominent role. High definition (HD) sets are currently widely available and even quad-HD panels are already being introduced. Although HD video transmissions are growing rapidly, legacy standard definition (SD) video will remain common in the near future. To display this legacy SD video on HD resolution and beyond, high quality spatial up-scaling techniques become increasingly important to obtain optimal picture quality and fully benefit from the high potential of high-resolution displays. Various up-scaling techniques are available, varying from linear1,2 to advanced non-linear3,4 techniques. This paper focuses on improvements of an accurate non-linear method, which is structure adaptive4. The method consists of classifying each pixel into a specific structure class, using adaptive dynamic range coding (ADRC), and applying an optimized interpolation filter within each structure class. Classification by ADRC is both simple and effective, but may be improved in terms of efficiency and even in accuracy. A more efficient and accurate classification would improve the accuracy and memory efficiency of the interpolation as a whole. In this paper, an alternative structure classification based on so-called principle components analysis (PCA) is considered: PCADRC. By performing the structure classification on a statistically optimal representation, rather than on the pixel values directly, one could expect the classification to be more optimal. Furthermore, the principal components based representation is inherently suitable for a ‘lossy’ class reduction, making it a memory efficient classification. II. STRUCTURE-ADAPTIVE UP-SCALING Content-adaptive up-scaling is characterized by adapting the filter coefficients to the local image content. The adaptation, or classification of the local image content, assigns a class to specific image characteristic(s). The most common image characteristic is the local image structure: both ADRC as well as PC-ADRC operate exclusively on the local image structure. A. ADRC classification In this classification, each pixel value in an aperture is compared with the average pixel value of that aperture. If the pixel value is equal or smaller to that average, ADRC assigns 0 to the pixel, otherwise it assigns 1. For example, a 3x3 aperture would generate 9 bits, which are combined into an integer value, which represents the class number. The number of classes and thus the LUT depends on the size of the aperture. Class redundancy is potentially large: the number of classes increase exponentially with the number of pixels in the aperture. Furthermore, this will be, in general, not the statistically most optimal way of classification, which may not result in the highest possible accuracy. B. Principal-Components based ADRC (PC-ADRC) This classification method is similar to the ADRC method, but operates on an alternative representation of the pixel values in the aperture. Rather than N pixel values, this new representation exists of N coefficients that correspond to the N principal basis functions, or ‘components’. The basis functions are determined beforehand on a set of representational images. The first basisfunction is, on average, the most significant for reconstructing the original pixel values, whereas the Nth basisfunction is the least significant for image reconstruction. The classification consists of selecting the sign of the principal coefficients: - means 0, + means 1. This classification has two potential benefits. As the representation itself is derived from the image statistics, one may expect its most significant bit to be representative of the local structure. Moreover, preserving the ‘most important’ basis functions and ignoring the other basisfunctions results in a straightforward decrease of the LUT size. III. ANALYSIS The two classification methods, ADRC and PC-ADRC, were evaluated in terms of image quality and storage cost (size of LUT). The image quality was evaluated subjectively and objectively by measuring the MSE between the reconstructed and original HD. On average, objectively and subjectively PC-ADRC attained about the same accuracy as ADRC. Looking at cost reduction, PCADRC indeed reduced the LUT size significantly before becoming apparent on the subjective evaluation in contrast to the objective evaluation. IV. CONCLUSION Our PC-ADRC classifier confirms that ADRC is about near optimal and highly accurate. However, if memory cost is an issue, PC-ADRC can be a suitable scheme for reducing the size of the LUT by leaving out the least significant principal components without notably sacrificing image quality.

REFERENCES

**Image Pipeline Tuning for Digital Cameras 🌟**

Jianping Zhou (Texas Instruments, USA)

In recent years, digital camera products have grown explosively as digital photos and videos have become more and more popular in our daily life. Thanks to technology innovations, digital cameras are used in a variety of applications, such as digital still cameras, camcorders, camera phones and video surveillance. To produce digital images, a digital camera embeds complex digital signal processing in the image processor, which may be called an 'image pipeline.' The image pipeline generates a digital color image in real time from raw data produced by a camera sensor. Therefore, the image pipeline plays a key role in digital camera systems and a good image pipeline is crucial for high quality digital images. Besides the image pipeline, a digital camera system comprises a lens, camera module, and sensor. All components play a part in image quality and they interact with each other. Therefore, the image pipeline must be tuned for particular camera system components based on their characteristics. This is a challenging task. First, different lenses, camera modules and sensors have different characteristics. Second, there are no standard image quality metrics. Traditional reference-based image quality metrics cannot be applied here because the desired image in human perception is different from the actual scene captured by a digital camera. Third, different users have different preferences for image quality. Much has been published about a broad range of image pipeline algorithms, but little information is available about image pipeline tuning techniques. In this paper, we first present an overview of basic algorithms inside an image pipeline, such as white balance, color filter array interpolation, color correction and Gamma correction. Then we introduce how to tune the image pipeline for different sensors to achieve the best image quality.

Jianping Zhou is a Member of Technical Staff with the DSP Solution R&D Center, Texas Instruments. He is responsible for designing, developing, and optimizing video and image processing algorithms, architectures, and software for digital cameras and camera phones. From 2002 to 2005, he was a Research Assistant with the Coordinated Science Laboratory, University of Illinois. From 2000 to 2002, he was working at the Internet Media Group, and then Wireless and Networking Group, Microsoft Research Asia. From 1997 to 2000, he was a Research Assistant with the National Laboratory on Machine Perception, China. He received the Ph.D. degree in electrical engineering from the University of Illinois at Urbana-Champaign. His research interests include wavelets, computer vision, digital color processing, image and multidimensional signal processing, multimedia coding and communication, and multimedia delivery over wireless Internet.

**Enabling Technology 3 🌟**

Room: Trinity 4

**Using Serial Resistors to Reduce the Power Consumption of Resistive Touch Panels 🌟**

Y. Bai (Fu Jen Catholic University, Taiwan); Chang-Yu Chen (Fu Jen Catholic University, Taiwan)

Touch panels are categorized into capacitive, surface wave, optical and resistive. The advantages of the resistive touch panel are that it is thinner, has a higher resolution, lower power consumption and a lower price, thus it has come to account for 75% of the global market in touch panels. In this paper we analyze the characteristics of the resistive touch panel to find a simple way to reduce power consumption. The resistive touch panel must be calibrated before use. The A/D converter of a panel receives both the maximum and the minimum of analog voltage. The analog voltage will be determined from the translation of X or Y coordinates. After the analog voltage is input it encodes 12bit digital data for the representation of the touch location. Commonly the touch panel can have a resolution of 4096*4096 which locations can be stored in the register. We insert extra serial resistors to reduce power consumption at the sensing circuit of the touch panel circuit. These serial resistors change the detected analog voltage which will be calibrated again when the touch panel is turned on next time. If the value of the serial resistors is increased, the current consumption is decreased. However, after the calibration for the analog voltage the touch positions are not influenced, but both the representation format analysis and the coding range of the analog voltage are compressed into a smaller range, and the step size of the comparator of the A/D converter is increased when we insert the serial resistors. An analog voltage can be confused between two different codes if we insert very large serial resistors. In the design we measure the contact resistance of the original touch panel which is 800ohms and the current consumption which is 6.25mA. When we insert a serial resistor, 300ohms at each end, the total contact resistance is 1400ohms. For a four-wire resistive touch panel the current consumption is reduced from 6.25mA to 2.89mA which improves the current consumption efficiency by about 54%. For a five-wire resistive touch panel the current consumption is reduced from 15mA to 4mA. Long-term measurement of the error rate of the touch panel shows that there is a very minor increase in the error rate, from 1% to 2%, when we insert serial resistors from 0ohms to 300ohms. When a resistive touch panel has not been
touched, the current consumption is less than $1\mu A$, thus the touch panel consumes limited power. For long-term operation of a touch panel there is an average of 2.5% to 5% current consumption with respect to that of a whole PDA based on low to medium-frequency use of the touch panel. Hence, by inserting the serial resistors one reduces the current consumption by about 1.35% (2.5% 54%) to 2.7% (5% 54%) for a typical PDA.

Ying-Wen Bai is a professor in the Department of Electronic Engineering at Fu-Jen Catholic University, Taiwan. His research focuses on mobile computing and microcomputer system design. Ying-Wen Bai obtained his M.S. and Ph.D. degrees in electrical engineering from Columbia University, New York, in 1991 and 1993, respectively. Between 1993 and 1995, he worked at the Institute for Information Industry, Taiwan.

**inter-profile J2ME applications development and porting**

Yevgeny Knutov (N/A, Russia); Sergey Matetski (N/A, Russia)

Article describes basic principles and approaches for creating Java applications compatible both with DoJa and MIDP profile. These approaches take up porting applications designed for one profile to another. It considers basic areas of overlapping and different profile functionality. Provides guidelines for developers involved in J2ME application creation, porting and maintenance.

**Design and Implementation of an Instantaneous Turning-on Mechanism for PCs**

Y. Bai (Fu Jen Catholic University, Taiwan); Huang-Te Hsu (Fu Jen Catholic University, Taiwan)

Lately the “digital home” has created a need for many electronic home appliances. Some of them are redesigned from PCs and make the physical PC invisible everywhere in our living environment. Thus “system standby all the time” has become an important feature for turning a PC on instantly. When a PC is turned on, the system takes about 40 to 90 seconds for the operating system (OS) to be loaded. There are several kinds of method for shortening such long loading or booting time. First, we build up the RAID (Redundant Array of Independent Disks) to increase the throughput of the hard disk, or we add more cache memory on the hard disk. But we see no significant improvement in shortening the loading time while we are booting up a system due to the interface throughput limitation. Second, we may be able to shorten the loading time by using “Suspended to RAM” instead of “Turning off system physically” as well. All the relevant OS data could be saved into the system memory while we pretend to turn off the system, and these data could be reloaded directly when turning on the system. This paper considers using the “Suspended to RAM” method to replace the “Turning off system physically” method to perform instantaneously turning-on of the system just like other current consumer electronics have done. Because the OS has been stored in the DRAM, extra standby power is needed to keep the OS data on standby and ready to be reloaded. However, the cost of energy is constantly soaring, and according to “ENERGY STAR(R) Program Requirements for Computers”, the standby power consumption of a PC should be less than 1 watt. However, our measurement shows the standby power consumption to be about 2.4 watts, much more than it should be. In this paper, we redesign the circuit to reduce the standby power consumption of a PC. First, we use CMOS as the current switch, thus avoiding the static current consumption between chips and turning off the power supply of the network controller with the waking up LAN function. Second, we turn off the power supply of USB and PS2 by using the CMOS switch with the keyboard waking up function. Third, we turn off the power supply of the termination resistor for memory modules. Fourth, we use the high-efficiency pulse width modulation power regulator to replace the linear power regulator. Fifth, we enlarge the value of the divider resistors for the reference voltage to reduce leakage current. Overall, these improvements reduce the standby power consumption to 0.8 watts which meets the ENERGY STAR(R) Organization’s definition. In addition our design shows that the PC can be turned on within 3 seconds by reloading the OS in the DRAM.

Ying-Wen Bai is a professor in the Department of Electronic Engineering at Fu-Jen Catholic University, Taiwan. His research focuses on mobile computing and microcomputer system design. Ying-Wen Bai obtained his M.S. and Ph.D. degrees in electrical engineering from Columbia University, New York, in 1991 and 1993, respectively. Between 1993 and 1995, he worked at the Institute for Information Industry, Taiwan.

**Efficient Signal Conditioning For Microcontroller Based Medical Solutions**

Murugavel Raju (Texas Instruments, USA); Kripasagar Venkat (Texas Instruments Inc., USA)

Signal conditioning is an important aspect in almost all analog measurement applications such as Blood pressure monitors, Heart Rate monitors, Digital hearing aids, Magnetic Resonance Imaging, Pulse Oximeters etc. While some of the high end medical applications need Digital signal processors (DSP) for processing their complex algorithms most portable medical applications need only a Microcontroller. Power consumption is of primary interest for portable battery operated Medical equipments. The use of Microcontrollers [3] boasts of lower power consumption compared to DSP allowing extended battery life, but has limited processing capabilities compared to a DSP. Filtering is the fundamental aspect of any signal conditioning scheme and the choice of digital filters over analog filters has several proven advantages. Digital filters when implemented on fixed-point machines gives rise to overflow, round-off noise and finite word-length effects [2]. This can be minimized by increasing the number of bits to represent the filter coefficients. In almost all cases the filter coefficients would be floating point numbers and.
have to be scaled to fixed-point numbers [1]. This potentially leads to a loss of coefficients if their dynamic range is huge. Additionally if the microcontroller does not have a hardware multiplier the CPU bandwidth required for would be high resulting in high power consumption. This paper proposes an efficient way to implement digital filters on a microcontroller without the need to scale the filter coefficients. The method uses an extension of Horner’s method to support an integer-float multiply. This method exhibits excellent accuracy when compared to implementations using only floating point math [4]. Horner’s method does the multiply and accumulate using only single cycle shift and add operations when used with certain microcontrollers such as Texas Instruments MSP430 thus enabling a fast multiply. The algorithm is based on the position of the binary 1’s and their difference in bit positions to adjacent 1’s. Floating point multiplication is achieved by extending this concept to the fractional part and the integer part. For this method to be functional the multipliers (filter coefficients) must be known in advance, which is the case for the filter coefficients. The only trade off using this method is increased code size because each multiplication has to have a dedicated software routine. The use of the Horner’s method in conjunction with Canonical Signed Digit (CSD) [1] number representation further reduces the CPU bandwidth required for each multiply. Example of IIR and FIR filters has been shown using this method on Texas Instruments’ ultra low-power MSP430 family of microcontrollers [3]. The analog signals digitized by an on-chip 12-bit Analog to Digital converter is used as the input at each stage. Performance in terms of frequency response, CPU cycles and code size has been discussed. The memory requirement is based on the filter order, and the number of bits chosen to represent the coefficients. The approximate increase in CPU bandwidth for every increase in order is about 35 CPU cycles. This is approximately ten times faster in comparison to a C library integer-float implementation and at least four times faster than the conventional integer-integer multiplication algorithms [4]. Simulation results include the implementation of a simple IIR notch filter rejecting the 60 Hz hum and FIR filters that exhibit narrow band responses, which are highly desirable for signal conditioning of medical analog signals. Performance figures depicting the frequency response of filters using this method in comparison to a floating point implementation show excellent performance improvement. References 1. Hewlitt, R.M.; Swartzlantler, E.S., Jr., “Canonical signed digit representation for FIR digital filters”, SiPS 2000. 2000 IEEE Workshop on Signal Processing Systems, 2000, 11-13 Oct. 2000 Page(s):416 – 426. 2. S. K. Mitra, “ Digital Signal Processing, A computer based approach ”, 2nd Ed. Mc Graw Hill 2001. 3. Texas Instruments, MSP430 family of microcontrollers, www.msp430.com. 4. Venkat, Kripasagar, “Efficient Multiplication and Division Using MSP430”, literature number SLAA329, Texas Instruments Inc.

Kripasagar Venkat is an Applications Engineer for the MSP430 Microcontrollers at Texas Instruments. He has a Master of Science degree in Electrical Engineering from University of Texas at Dallas with an emphasis on Digital Signal Processing.

**Low Cost Testing of Low Power Wireless Transceivers**

Larry Zhang (Texas Instruments, Inc, USA); Paal Reichelt (Texas Instruments, Inc., USA)

Parallel low cost testing of low power wireless RF transceivers is realized using a low cost RF mixed signal tester. Special design for test (DFT) considerations for the transceiver has made the implementation of low cost multiverse RF testing possible with reduced test time and test development effort and improved throughput. Careful load board design considerations have reduced load board cross talk, signal noise coupling, and susceptibility for electromagnetic interference.

**Architectural Design of Multiplicative type-II Fuzzy Cellular Neural Network in CMOS Technology**

Jui-Lin Lai (National United University, Taiwan); Rong-Jian Chen (National United University, Taiwan); Cheng-Fang Tai (National United University, Taiwan)

The architecture of multiplicative type-II Cellular Neural Networks with Fuzzy (FCNN) is proposed, which is with local connectivity advantageous suitable implemented for VLSI. The fuzzy AND and fuzzy OR morphological operators in the Type II FCNN can be operated by Min and Max operations for the feel forward template from neighbouring cell of input image in dynamic parallel processing. Base on the proposed FCNN structure that the neuron, Min/Max, analog multiplier, transfer and control circuits are adopted and verified by the HSPICE. The desired experimental chip of 3×3 multiplicative type-II FCNN is fabricated using TSMC 0.35μm 2P4M CMOS technology. The measurement results have successfully verified the correct function of FCNN. There have a great potential in the VLSI implementation of neural network systems for binary and gray-level patterns in image-processing for consume electronic applications.

Jui-Lin Lai (M’01, SM’04) received the B.S degree from the Electronic Engineering, National Taiwan University of Science and Technology, Taipei, Taiwan, 1984. He received the M.S and Ph.D degree in the Institute of Control Engineering and the Institute of Electronic Engineering from the National Chiao-Tung University, Taiwan, R.O.C., in 1990 and 2004, respectively. In 1984, he joined the faculty of Electronics Engineering Department, National Lien-Ho Institute of Technology, where, in 2003, the Institute was upgraded and renamed as the National United University. He currently is the Associative Professor in this University. His research interests are in analog VLSI design, neural networks, and computer architecture.
Workshop 1

Building an Automated Test System in 3 Hours

Room: Trinity 8

3:15 PM - 5:00 PM

Audio Video Technology 4

Room: Trinity 6-7

Fractal Dimension Analysis of Audio Signals for Musical Instrument Identification

Shanmugam Gunasekaran (Tata Elxsi Limited, India)

Automatic sound source recognition plays an important role in developing automatic indexing and database retrieval applications. Recognition of musical instruments in multi-instrumental, polyphonic music is a difficult task, which is yet far from being achieved. Efficient instrument recognition techniques in solos can help to deal with this task. The application of fractal geometry to audio signals and musical instrument recognition system is not been widely experimented. A very important characteristic of fractals, useful for their description and classification, is their fractal dimension D. The fractal dimension provides an objective means of quantifying the fractal property of an object observed in the natural world. This paper introduces an instrument identification system based on fractal dimension analysis of audio signals. This research is organized in four parts. First part of this research investigates fractal-dimension segmentation for feature extraction and recognition of isolated musical sounds. Dynamic Time Warping (DTW) method is used to match the sampled music with existing templates. The fractal dimensions of the synthesized fractal signal were then evaluated using the power spectrum method. Second part of this research explores extraction of conventional as well fractal-dimension based feature set from the segmented musical signal. The conventional feature set of the proposed system has very extensive collection of 91 features, which includes spectral, temporal, harmonic, perceptual and statistical features of the musical signal. The application of fractal theory depends on the accurate measurement of D, which is essential in fractal-based instrument recognition. The fractal-dimension D is used to make the proposed instrument recognition system more robust and accurate. Third part of this research introduces a novel feature selection algorithm, which identifies the most essential features from the set of all features extracted. The rest of the features were not used in the classification process. The feature selection algorithm works with the initial knowledge that has been attained from the fractal dimension D. This feature selection algorithm enables the system to work on real-time signal. But a minimum compromise on performance is assumed. The proposed system is designed in such a way that the feature selection algorithm could be optional and used if the system runs with critical resources. Fourth part of this research describes about the neural net classifiers. The system has been trained and tested with three major neural net classifiers (Generalized Linear Model, Multilayer Perceptron, and k-Nearest Neighbor). The system has been trained and tested for a set of 12 Indian musical instruments. All musical samples used were solo instrumental recordings. The samples were recorded from musical instruments and are not synthesized sound samples. The instrument selection has been made based on the family of musical instruments that it belongs to. Three Instruments from every family has been chosen. This proposed system has shown better results while comparing with other conventional instrument recognition systems. There was little variance in performance among the three classifiers. The results with different neural net classifiers have shown k-NN classifier as the best suitable one for this proposed instrument recognition system.

GPU-based Background Illumination Correction for Blue Screen Matting

Nicolas Ley (Technical University of Ilmenau, Germany); Christian Weigel (Technische Universität Ilmenau, Germany); Markus Mehnert (Technische Universität Ilmenau, Germany)

Separation of foreground objects from an almost constant backing color for video applications is still a common problem. For non-realtime situations there is a wide variety of different powerful mathematical approaches that can deal with most of the mattig problems. For SD/HD studio realtime keyers most solutions are not applicable due to their algorithm complexity or high effort in user interaction. Excellent hardware keyers, such as UltimatteTM work on most occasions, but even under controlled lighting in a blue-/greenscreen matting problems may occur, or creativity is limited by necessary lighting conditions. As a preprocessing algorithm for traditional chroma keying systems, we present a simple background illumination correction based approach for improving matting problems with uneven or poor lit blue-/greenscreens. Using the computational power of GPU computing the presented algorithm is realtime capable and offers an improvement for achievable mattes quality. The lighting of the background plays an important role how mattes will turn out satisfactory. Poorly lit blue-/greenscreens degrade the
quality of the matte that can be extracted. Uneven lighting is a common problem, because it is very difficult to get the lighting both bright and uniform across a broad surface. An other frequent problem is a too bright or too dark lit background combined with uneven lighting. Pulling a matte without high cleanup level is nearly impossible. The presented preprocessing algorithm consists of several buildings blocks, nevertheless the method is clear and efficient. The necessary steps for creating a corrected image with the computation using the GPU is outlined in the following enumeration: 1. download image to the GPU 2. subsample image 3. pull matte with simple chroma key with high cleanup 4. dilate the matte 5. automatic inpainting 6. gaussian convolution 7. upsample image 8. composite illumination corrected image 9. readback image or pull matte with a shaderprogram After downloading the image to GPU's memory represented as texture, the first step makes use of the assumption that the illumination change is low-frequency. Therefore a subsampling of the image is possible, the low frequencies still remain. Dependent on the subsampling factor (subsampling by a factor of max. 4 is proposed) not only matte pulling and dilatation computation time benefit, furthermore smaller gauss kernels can be used. As the channel for background estimation either luminance or the blue/green channel is used. A simple chroma key matte can be computed by channel per pixel min-max-Operations of the input image, resulting in a binary matte. A simple dilate operation is implemented as a filter kernel, the size depends on the subsampling level. After this operations foreground elements are determined. Foreground mask and image are now used for a simple inpainting algorithm. As presented the inpainted image can be used for illumination estimation by convolving with a large Gaussian kernel. The optimal size of the gaussian filter is dependent upon the scale of the objects in the image and the overall size. Experiments have shown that as a starting value 1/20 of the image width is recommended. The illumination estimation image is the resampled and the correction is done by mixing the original image with our estimation. The algorithm presented in this abstract implements an easy preprocessing method for improving the keying result for images with uneven lighting in the backing. The sample implementation is done on the GPU with common shaderprograms and needs no readback during computation. With the illumination correction as preprocessing there is more headroom in the creativity for lighting situations in live productions, the “poor-lit-bluescreen-problem” gets a little less problematic.

Implementation of Interactive Multi-view Visual Contents Player based on MPEG-4 Systems ★
MyungSeok Ki (ETRI, Korea); Injae Lee (ETRI, Korea); Kyungae Moon (ETRI, Korea); Jin Soo Choi (ETRI, Korea)
1. Abstract In this paper we propose an Interactive Multi-view Visual Contents Player for Multi-view A/V. that is composed of panoramic, multi-view video/image, audio and the other media data. In spite of advantage multi-view video can supply more wide and various information for general rectangle video, it’s can’t used widely, because composition method and format of Multi-view contents can differs according to a author and it is very limited to express composition information, and such limited characteristic affects to Playing that. Furthermore to make multi-view content audio and video synchronized each other temporal and spatially is need to special system, but currently standard for multi-view content is insufficient. International standard MPEG-4 is includes not only A/V compression but also scene description method to compose contents using media and which defined nodes for 2D and 3D contents to express scene. And it’s includes that scene composition method using audio and extra data in addition to image. An Interactive Multi-view Visual content player is based on MPEG-4 Systems Decoder, and to authoring content we using 2D and 3D nodes of MPEG-4 Scene descriptor profile. Player support that view change of Panoramic/Multi view video, surrounding audio effect with synchronized video and interactive data service. If we produce contents by proposed method and play it, player can be adapted flexibly even if changed that acquisition environment or authoring condition. And if using this player we expect to various interactive data service as well as multi-view service. 2. Feature of Interactive Visual Content player -Panoramic and Multi-view visual content playing -MP4 file supporting based on MPEG-4 Systems standard -Panoramic scene navigation and surround audio effecting adapted video -Provide interactive functionality using JPEG, MNG, Graphic object and the other media objects

Emotion-based Textile Indexing System using Pattern Recognition ★
Na Yeon Kim (Konkuk University, Korea); Eun Yi Kim (Konkuk University, Korea)
For a given product or object, predicting human emotions is very important in many business, scientific and engineering applications. In particular, the emotion-based textile indexing has been considerable attention, as it can be applicable to the E-business and furthermore help pattern designer. In current, the textiles are manually annotated by human experts in the current, which cause a huge amount of time and effort. To reduce the cost and time, automatic indexing system should be developed to classify the textiles based on the emotional features. However, it is difficult to directly predict the human emotion from the textiles, due to the ambiguity of human emotion. Therefore, it is an important issue to find the correlation between the human emotion and physical features such as color, texture and shape information included in the textile images. Related to these issues, some works have been investigated. Although they showed the correlation between some physical features and human emotions, they did not provide the automated system to extract the physical features from the textiles and analyze

http://edas.info/showProgram.php?_of=showProgram=&c=5136&form=bio%5D=1&program_view=everybody&action=Show+conference+program
the features. In this paper, an automatic system is developed for emotion based textile indexing. Our indexing system labels an input textile using ten pairs of adverse emotional features expressed as adjective words: {romantic/unromantic, clear/unclear, natural/unnatural, casual/uncasual, elegant/inelegant, chic/unchic, dynamic/static, classic/non-classic, dandy/non-dandy, modern/non-modern}. The basic assumption of the proposed approach has that there would be the correlative relationships between the human emotion and the pattern in image. Therefore, we collected total 220 textile images from Pattern-Book, and then, conduct the survey to investigate the correlation between the emotion and the pattern on 20 peoples. The result shows that a human emotion is deeply affected by the certain pattern. Based on the relationships between the human emotion and the pattern, we construct the neural network (NN)-based indexing system to predict the human emotions from the give textile images. The proposed indexing system is composed of feature extraction and classification. To describe the pattern information in the textiles, the wavelet transform is used. And the neural network is used as the classifier. Wavelet transformed data extraction scheme (WTDES) uses 1st, 2nd, 3rd momentums of each sub-blocks extracted from 6-levels wavelet transform in 64x64 input image, and these values are input to the NN. The NN is composed of total 72 input nodes and 1 output node, and the number of hidden nodes is determined by the experiment. To assess the validity of the proposed indexing system, it was applied to recognize the human emotions in textiles. For the respective textile images, twenty peoples were selected to manually annotate according to the emotions that they feels from the images. Then 120 images of 220 collected images were used for training the NNs and the other was used for test. Our system produced the precision of 98% and the recall of 90% on average. This result confirmed that our system has the potential to be applied for various applications such as textile industry and e-business.

**Automotive Entertainment & Information**

Room: Trinity 5

**Reliability and Scalability of the Kilavi Building Control Network**

Mikael Soini (Tampere University of Technology, Finland); Lauri Sydanheimo (Tampere University of Technology, Rauma Lab, Finland); Markku Kivilkoski (Tampere University of Technology, Finland)

Wireless sensor networks are based on physically small and resource-constrained nodes exchanging environment or application related information with each other. These networks can be utilized in wide area of applications such as home control, structure surveillance, medical monitoring, forest fire protection, and wild life observation. Sensor network applications related to home and building environment and thus to the consumers are amongst the most interesting and feasible in this field. Building is, however, very challenging environment to implement a reliable wireless sensor network. Though sensor nodes usually have very low mobility, there can be changes in sensor network topology caused by changing environment (variable floor plans, furniture replacements, closed doors etc.). These variations in the environment affect significantly to radio frequency propagation that has strong and hard-to-predict path loss even in stable building environment because of different construction structures and materials. These issues are taken into consideration in Kilavi based building control. Kilavi protocol is developed for low-power and low data rate wireless communication in a building environment. This protocol offers a wireless communication interface for all kinds of control and measurement devices in this area. Kilavi is based on centralized master/slave architecture, multihop communication, very short low overhead packets, and fairly low 433MHz communication frequency. Earlier work has indicated that Kilavi enables long lifetimes for power critical network leaf nodes and simplifies security architecture with end-to-end keys. In addition to this, results have been shown that Kilavi has reduced packet overhead in device discovery and network maintenance, and routing is very efficient compared to some distributed solutions. The utilization of high capacity nodes as cluster heads installed, for example, to power outlets can be used to increase network range and communication reliability between resource-constrained sensor nodes and the network master node. In Kilavi, these cluster-heads operate as data forwarders to simplify and improve network operation. Earlier packet loss and communication range measurements tested the reliability of different shared channel reservation schemes. Simple RTS/CTS scheme provides the basis to reliable communication with reasonable packet overhead and thus power consumption. These measurements were used as a foundation to the simulations that test how Kilavi operates in different radio (data rate and modulation), network (hop count, number of sensors and cluster heads), protocol (packet length) and traffic (data packet transmission interval) conditions. The measure of communication reliability is end-to-end packet loss. Simulations include both single- (star topology) and multihop (cluster-tree topology) networks. Simulation results indicate that single-hop Kilavi network has quite limited sensor capacity but with clustering and multihopping a reliable network that scales quite well to several branches and multiple hops can be obtained. The full paper analyzes the simulation results thoroughly. Functional wireless sensor network is a vital component that provides the ambient intelligence for integrated smart home implementation that includes home multimedia and high-speed communication systems. As an example, sensor network can determine conditions and occupant preferences inside the house and thus seamlessly set ideal conditions when tenants are watching a movie or when
they are absent.

Mikael Soini was born in Finland in 1979. He received his M.Sc. degree in electrical engineering from Tampere University of Technology (TUT), in 2002. He is currently a Research Scientist at TUT Institute of Electronics, Rauma Research Unit and works as a manager of Wireless Automated Building Systems research group. He is currently finishing his Ph.D. thesis which focuses on developing wireless control and monitoring reliability and usability in a building environment. His research interests are focused in the area of wireless communication, sensor networks, home&building control systems.

A Modular Framework for Applications Development on Embedded Platforms ★

Nicholas Vun (Nanyang Technological University, Singapore); Kelvin Xu (Nanyang Technological University, Singapore); Zhirong Foo (Nanyang Technological University, Singapore)

While typical embedded system used to be only designed for a specific application, keen competition in consumer electronics market will require a product to be designed such that it can be easily upgrade with new and noble features to provide product differentiation among similar applications. This is now feasible due to the availability of the latest embedded system platforms where choice of computer processing power is not as limited as before. Advancements in embedded platform technologies have approached the level similar to desktop personal computer (PC) system such that when installed with appropriate operating system and with careful program design, complex applications that used to be only executable on PC can also now be run on the embedded platform. A good example of such a product is the automotive dashboard console that previously only requires to provide basic functions like radio and CD playback. However, latest console design provides many more features. Examples includes touch screen panel that display map for directory/navigation functions, GPS receiver, Bluetooth connectivity for digital music streaming from music player (e.g. iPod), hands-free function that work seamlessly with mobile phone, car alarm system that incorporate remote monitoring and control through cell phone infrastructure. As more and more functions are to be incorporated into the embedded system, conventional mean of developing the application that incorporate all the features within one program will be laborious, hard to debug and maintain. With the frequent feature enhancement prevalent in the consumer electronics product, feature upgrade and new function addition will also require the modification of existing modules. Without proper system design, this process will be painstaking, or even worse, affect the performance of the existing system and delay the rollout of the product. This paper describes the framework that is used to develop an embedded system to avoid the pitfalls of such scenario. The embedded system described in the paper is developed for automotive dashboard console. The system is based on an ARM9 processor running a ported version of Linux 2.6 o/s with interface through a touch screen LCD display. The system features a number of advanced functions which include a database driven digital map system for in-vehicle navigation and directory purposes, wireless 802.11g connectivity, Bluetooth connectivity with Personal Area Network (PAN) profile and Advanced Audio Distribution Profile (A2DP), and an interface to a Bluetooth GPS receiver. Other functions currently under development include Bluetooth’s Hands-Free Profile (HFP), Phone Book Access Profile (PBAP) and wireless ad-hoc network for inter-vehicle communication. Multi threading is probably the most common method of implementing a system that needs to provide multiple functions. However, during the process of developing the system described in the paper, it becomes obvious that as the system get increasingly blotted with features, developing and debugging of the system based on multi-threading alone will become very time consuming, unwieldy and laborious. Furthermore, future features enhancement for the system will also involve careful understanding, reviewing and modification of existing codes. To overcome this limitation and restriction, a modular framework is adapted in the design by coding each function as an independent program. Each program will be run as separate process upon execution. For process that needs to exchange data with other process(es), network socket based on IP address and port number is used to provide client-server like communication channel, where the server waits for incoming client request by listening to a specific port. To conceal the underlying modular structure of the system from the user, a graphic Launcher program is used to seamlessly ‘integrate’ all the programs and provide a single point of interface for the user to interact with the system. The Launcher provides touch screen button icons that user can use to launch each function (which in fact is a program). Launcher also performs the synchronization function among the different programs when needed. E.g. the system is programmed to automatically terminate the map directory program when the navigation program is selected. The paper will compare the various techniques for inter-process communication, and describe in detail the modular framework used in developing the automotive embedded applications based primarily on socket, with multithreading support as needed. Examples of the implementations (e.g. the Launcher program) will be presented to illustrate the benefits of such an approach.

An Enhanced Multi-Path Scheme for QoS Guarantee in Wireless Sensor Network ★

Oh Hyun-Woo (Electronics & Telecommunications Research Institute, Korea); Intak Han (Electronics and Telecommunications Research Institute, Korea); Kwang-Roh Park (ETRI, Korea); S.h Kim (Chungnam National University, Korea)

The wireless sensor network consisting of excellently many sensors is mainly used for the application performing...
the mission such as target tracking and emergency response more than the application respecting the general data communications. This application demands the sensing data transmission of a real-time and the exact transmission of sensing data. When sensed data were received with a sink, the real time data transmission has the restriction of the time for having the validity. Even though sensed data undergoes some delay, it can have the validity with a sink and not be effective. According to a situation, it can accommodate the delay which in some cases does not exceed the threshold level. The transmission pattern of the sensor monitoring a temperature and it of the sensor for grasping the emergency situation are different. Therefore, in the wireless sensor network, it has to be satisfied the requirement having the different time restriction according to a situation in other words according to an application. Particularly, it has to satisfy not only the time restriction which has to be delivered during timeline but also the reliability which has to be accurately delivered without the loss of data in the case like an emergency or real time. In this paper, we propose the scheme which comprises a multi-path on the wireless sensor network in order to guarantee various QoS requirements as follows. - Service differentiation: a service is discriminated on the wired network like the multimedia streaming service, the real time voice service and data service. In this way, there exists the differencing of a cycle and the amount transmitting data sensed in the wireless sensor network according to a service. - Time restriction: even though it says to be data sensed in the identical sensor, it can have the limitation nature of the different time according to a situation like an emergency. - Transmission reliability: as to the mobile network, the transmission rate remarkably falls down in comparison with the wired network because the propagation characteristic and the loss. Data sensed in the wireless sensor network requires the transmission of exact data according to an application. Particularly, in an emergency and battlefield, the transmission of exact data is essential. In this paper, we propose the multi-path scheme which utilizes the virtual grid in order to be satisfied this kind of requirements. The method using the size and transfer period of sensing data in order to select a route on the multi-path according to a service for the service differentiation is proposed. The proposed algorithm dynamically selects an alternative path besides the route which already selects according to a situation on a multi-path and it gives the priority about the shortest path to sensing data which have more strict time restriction. Finally, our scheme guarantees the transmission reliability as transmitting the sensing data through two or more routes by using the dualized multi-path on the virtual grid. In this paper, we introduce some realistic scenario, propose a model and proof the improvement of a performance for our scheme through the mathematical analysis in comparison with the existing SPEED and MMSPEED.

An Algorithm for Congestion Control and Routing based Context-Aware in Wireless Sensor Networks

Oh Hyun-Woo (Electronics & Telecommunications Research Institute, Korea); Intak Han (Electronics and Telecommunications Research Institute, Korea); Kwang-Roh Park (ETRI, Korea); S.h Kim (Chungnam National University, Korea)

The wireless sensor network can be comprised of excellently many sensors. That is mainly used in order to perform the mission which is particular than the general data transmission. This application considers the environment like a battlefield, a disaster, an emergency and the risk exploration. In this environment, the behaviour in which a user or an administrator intends can be made but the situation which does not intend at all can occur. For example, there is the temperature sensor which it sets up in the surrounding of expressway in order to analyze the ecosystem of the freeway neighboring. These periodically transmit sensing data in the normal times about the gradual temperature change. The vehicle collision published on a freeway. The fire comes as a result. If it is the case, these randomly transmit sensing data about the sudden temperature change. Normal, because the radio range of the radio frequency sensor is small to the picocell less than, an affect by the outbreak situation senses in many sensors. As to these, a collision is generated with sensors which broadcast sensing data and a neighbor do. In conclusion, as to the wireless sensor network, the overcrowding of sensing data is generated with the unforeseen situation. The special mission can be performed to the wireless sensor network itself. However, it operates with for the more effective application with another network like an internet. In the wireless sensor network, the gateway which connects the wireless sensor network and internet so that here and there sensed data can be assimilated to the application connected to an internet if an overcrowding is generated is passed through. The congested situation generated in the wireless sensor network can induce the congestion in a gateway. In this paper, we propose the congestion control algorithm using the dynamic multi-path in the wireless sensor network and the congested situation processing algorithm using the storage. We distinguish the congested situation into two stages. First, it is the case of drastically transmitting sensing data with the situation which is exceptional in the wireless sensor network in sensors. Another one is the overcrowding generated in a gateway with sensing data which it randomly transmits in the different location in sensors existing. We consider the some kind premise in designing the congested situation control algorithm. - The congested situation recognition: the congested situation is classified in three point of views. Firstly, the source which produces sensing data and which it transmits can sense the congested situation. Secondly, the intermediate node existing in the transmission path of sensing data can grasp the congested situation. Thirdly, the gateway connecting the wireless sensor network and internet can recognize clearly the congested situation. - The load balancing according to the service differentiation: the size of
data which a source transmits in the normal times and size of data which it has to transmit in the outbreak situation can be different. As to the intermediate node existing in the transmission path, a source can transmit other sensing data through the different route. -The transmission reliability through the route duplexing: by using the virtual grid, a source sets up the transmission path and selects the route according to the service differentiation. The transmission path formed with the virtual grid has the structure of being dualized. -The storage base congestion control: the storage technology develops brilliant and has the storage micro sensors within. -The routing according to the situation recognition: data sensed in the outbreak situation may be routed to the separate application. Even though it says to be data sensed in the identical source, it can be routed according to a situation to the different destination location. We propose the algorithm controlling the congestion generated in consideration of conditions which the upper part mentions in the wireless sensor network with the outbreak situation. If data in which a source is sensed are collected, the virtual grid is produced. The route where it can transmit sensing data on the virtual grid is generated between a source and destination location as the grid lattice number. These routes have the symmetrical structure of being dualized. The arbitrary node dynamically selects the transmission path according to the size and transfer period of data which a self processes to the destination location if the congestion is recognized. If the sensor having the storage built-in receives a message, while it stores sensed data in the storage, decreases its own transfer rate. In this case, it is used for the application which demands the transmission of exact sensing data although it submits to the little bit of a delay.

Integrated Service-Oriented Home Management Systems ★

Wen-Hui Chen (National Taipei University of Technology, Taiwan)

Recently, Smart Environment has become a hot research topic and drawn many researchers into this area. For the living environment, people are seeking for a high-quality life style with convenient, comfortable, and secure living space. The great advances in technology and interdisciplinary collaboration have made it possible to build such an intelligent living space. From industry to academia, a lot of smart home projects and prototype systems have been conducted and continuously undertaken. One of the challenges to the smart home is the integration of heterogeneous devices for different applications. In this study, an integrated Service-Oriented Home Management System (SOHS) is proposed for providing home energy management, mobile security surveillance, and tele-homecare services based on multi-agent technology. The application of home energy management aims at reducing energy consumption by controlling home appliances efficiently. It is an important application and has gained more people’s attention after Kyoto Protocol came into effect. The air-conditioner is the most energy consumption device among home appliances. To control air-conditioners operating in a suitable temperature without affecting people’s comfort and avoid unnecessary energy consumption, an accurate thermal comfort model is required for controlling the air-conditioners automatically. In this study, the Adaptive Network-based Fuzzy Inference System (ANFIS) approach is adopted to build the thermal comfort model. In addition, a vision-based mobile robot is developed to meet the emerging needs of home security surveillance and tele-homecare services. The home robot features real time face recognition ability by using hybrid Principal Component Analysis (PCA) and Grey Relational Analysis (GRA) algorithm with 97.5% recognition rate. In robot navigation, an evolution algorithm and a fuzzy controller are designed to solve path planning and obstacle avoidance problems. For helping independent elders in urgent situations, a low cost wrist-worn alarm device is developed. The study also covers the technology used to create the proposed system with industrial standards.

A Novel Bidirectional TTI Application based on IPv6 using T-DMB ★

Youngho Jeong (Electronics and Telecommunications Research Institute, Korea); Soon-Choul Kim (Electronics and Telecommunications Research Institute, Korea); Chung Hyun Ahn (Electronics and Telecommunications Research Institute, Korea); Soo-In Lee (Electronics and Telecommunications Research Institute, Korea)

T-DMB (Terrestrial-Digital Multimedia Broadcasting) allows the consumers to view clear moving pictures in harsh reception conditions and also can provide an economical way of massive multimedia data services up to 1.7Mbps. It can provide a variety of data services such as traffic and travel information (TTI), electric program guide (EPG) and broadcast web site (BWS). Among these data services, TTI service has been spotlighted in aspects of economic influence and information usability. In order to providing TTI services, TPEG protocol was developed by EBU. However, it has been applied merely to two application areas. One is to transfer the road traffic status message (RTM) in case of an accident, out-of-door gathering, and so forth. The other is to convey the public transport information message (PTI) such as schedules and routes of buses, trains, flights, ships, and so forth. Recently, however, new needs of consumer for bidirectional TTI service become pronounced, that cannot be met through the existing RTM and PTI application. In this paper, we propose a novel bidirectional TTI application based on IPv6, TPEG-IPv6, which is satisfied to be interoperable with TPEG protocol, and verify the stability and effectiveness of that. The TPEG-IPv6 is designed to enable the users to receive real-time information of theirs' society in internet such as traffic, movie, show and event information through not wireless communication but T-DMB network. However, uploading for information to be transmitted through T-DMB is performed by wireless
Security & Digital Rights Management 1

Room: Trinity 4

E-Pedigree Discovery System and its Verification Service for Consumer’s mobile RFID Device

Juhan Kim (Electronics and Telecommunications Research Institute, Korea)

In this paper, we propose electronic pedigree (e-pedigree) discovery system (PDS) and its e-pedigree verification service for consumer's mobile device. The e-pedigree contains product information, transaction information, distributor information, recipient information, and signatures about an article that has a RFID tag through whole distribution. The aim is to try to reduce the introduction of counterfeit drugs into the legitimate pharmaceutical supply chain using Electronic Product Code (EPC) technology. The standard of the e-pedigree message has recently ratified by EPCglobal which is the non-profit organization charged with promoting the adoption of EPC technology. The PDS manages e-pedigrees which are including secured and trusted information that is generated, updated and digitally signed by the information servers of manufacturers, wholesalers, and retailers. The PDS provides securities such as access control, authentication, and privacy protection for the e-pedigree. It also supports an e-pedigree document query service and a verification service on a consumer’s mobile RFID device like a mobile phone or a PDA which can connect mobile RFID reader which is made especially for the mobile phone and the PDA. In this paper, we present the architecture of the PDS and how it works on mobile RFID environment. We also propose the structure of security functions of the PDS and the mobile device to support the e-pedigree verification service. Then we outline our pilot implementations such as a security library for the mobile RFID devices and mobile security gateway for the e-pedigree verification service including the PDS. Finally we end this contribution with concluding remarks.

Juhan Kim received his B.S degree from ChungNam National University, Daejeon, Korea, in 1997 and the M.S degrees in Computer Science from the same university 1999. He is currently a senior member of engineering staff at the Electronics and Telecommunications Research Institute (ETRI), Daejeon, Korea. Currently, his main research is focused on RFID and WSN security issues.

JavaCard-based User Key Management for IPTV Conditional Access Systems

Jinyoung Moon (Electronics and Telecommunication Research Institute(ETRI), Korea); Eui-Hyun Baik (ETRI, Korea)

A Conditional Access System (CAS) is essential to protection of the premium contents from unauthorized access in Pay-TV systems. To allow only authorized subscribers to view the contents, a CAS scrambles the contents with a scrambling key according to a scrambling algorithm. Generally, the scrambling key is also used to descramble the transmitted contents at a subscriber's set-top box. In order to securely send the scrambling key, the CAS transmits the scrambling key after encrypting it with another key. For security and efficiency, the CAS usually provides a multilevel key distribution scheme including the highest scrambling key and lower encryption keys. Traditional CA systems share the lowest encryption keys related to every subscriber by distributing them through smartcard to every subscriber or embedding them into set-top box because there is no way to securely share the initial encryption key in one-way broadcasting network. Therefore, the initial encryption key is fixed during the smartcard life cycle. However, IPTV, which a digital TV services through premium IP network, can distribute the initial encryption key out of band. In this paper, we proposed a JavaCard-based user key management technique
suitable for IPTV CAS. A user key is divided into a fixed user key and a variable user key. The fixed user key consists of a public fixed user key stored in the JavaCard and a private fixed user key stored in the subscriber management server. At the server side, a variable user key is generated periodically and is used to encrypt the other key. The variable user key is distributed to the subscriber through IP network only after being validated using his own JavaCard. We can judge it by comparing the asymmetric cryptographic computation. We adopted JavaCard instead of Smartcard because we can extend cryptographic computing power by installing new JavaCard program and can develop the JavaCard program in a hardware-independent way. The proposed user key management technique lengthens the available period of the fixed user key because the variable user key instead of the fixed user key is used to encrypt another encryption key. In addition, the proposed user key management technique extends the traditional CAS based on key distribution through CA without major modification.

**A Three – Party Authenticated Key Exchange Scheme Smartcard using Elliptic Curve Cryptosystem for Secure Key Exchange in Wireless Sensor Network**

Jongwon Yang (Kongju National University, Korea); JinMan Cho (ETRI, Korea); ChangHo Seo (Kongju National University, Korea)

As wireless sensor network has been the essential technology for ubiquitous computing, security issues are getting more attention from many researchers. Many security protocols for sensor network have been proposed. There schemes consists of symmetric key based ones and public key based ones. The security protocol that uses symmetric key such as SPIN requires many computing power and it is not suitable for resource limited sensor network. And they has poor scalability for enlarging network size. So research of security scheme using public key is current mainstream. Some public-key-based protocols such as TinyPK and EccM 2.0 have been proposed. And there are some third party authenticated key exchange scheme using validation information derived from password. These schemes exhibit poor performance and they are vulnerable to man-in-the-middle attacks. And the password-based Encrypted Key Exchange (EKE) scheme is weak to the attacks such as dictionary attack and message replay attack. In this paper, we describe the design of a protocol based on Elliptic Curve Cryptosystems (ECC) for securing key exchange in wireless sensor networks. The system includes password-based authenticated key exchange for providing secure communication service between users who use same server. This paper proposes a scheme using Elliptic Curve Differ-Hellman(ECDH) protocol for establishing shared key between nodes. This protocol use clustering to enhance the speed of key sharing. The proposed scheme use Elliptic Curve Digital Signature Algorithm (ECDSA) for minimizing key size and efficient broadcast authentication. Our scheme shows more efficient and secure key exchange.

**BATSMAN: a Byzantine Attack Tolerant Scheme for Mobile Ad hoc Networks**

Sajjad Pirahesh (Amirkabir University of Technology, Iran); Saeed Soltaniali (Amirkabir University of Technology, Iran); Salman Niksefat (Amirkabir University of Technology, Iran); Masoud Sabaei (Amirkabir University of Technology, Iran)

Ad hoc networks are collections of mobile nodes with links that are made or broken in an arbitrary way. They have no fixed infrastructure and, usually, have limited resources. The next generation of IT applications is expected to rely heavily on such networks. Deployment of wireless ad hoc networks has made some issues such as security and survivability of such networks more important than ever. One of the threats to these networks is Byzantine attack in which some of authenticated nodes, disrupt routing and forwarding of packets. In fact authentication mechanisms could not stop this type of attacks and one of the methods to detect them is to monitor nodes behavior. Recently some Byzantine attacks tracing protocols has been proposed. The most complete of them, combines a reliability metric based on passed history with an adaptive probing technique. All nodes must obtain the reliability information from the network by itself because it did not consider any trust to network entities due to nature of Byzantine attacks. In this paper we propose a trust model, based on human societies trust model and we define a trust parameter to networks which can discriminate miscellaneous networks, therefore protocol can behave dynamically in various types of networks. Furthermore, we have designed an information sharing mechanism based on network trust model which nodes could employ information gathered from other nodes and speed up their learning about network. This plan accelerates detection of malicious entities and improves selection of reliable paths and increase network’s survivability.

7:00 PM - 8:30 PM

**Special Session 1**

Graphical System Design for Academia
Room: Trinity 8

Thursday, Jun 21

8:30 AM - 9:30 AM

Keynote 3★

A Microcosm of Change: The Impact of the Videogame Industry on Social Networking and the Consumer Media Experience
Room: Trinity 1-4
Chair: Cyrus Cantrell (University of Texas at Dallas, USA)

9:30 AM - 10:30 AM

Audio Video Technology 5★

Room: Trinity 6-8
Chair: Jianping Zhou (Texas Instruments, USA)

Evaluation Method of Image Stabilizer for Digital Still Cameras ★
Kazuki Nishi (The University of Electro-Communications, Japan)
Various optical image stabilizers have been developed in many camera manufacturers for reducing camera-shake blurs, but they are not known exactly how effective they work for cancellation of camera-shakes. We propose a novel method for detecting the trajectory of camera rotations in three axis, i.e., pan, tilt and roll, from the captured image itself for examining quantitatively degrees of camera-shake or improvement by image stabilizer. The key idea consists of capturing the quickly switched test patterns that are displayed on a screen and calculating displacements between each pattern that is multiply recorded in the captured image while opening the shutter. The principle of the 3D camera-shake measurement and evaluation results of the camera-shake correction are presented.
He received the B.E. and M.E. degrees from the University of Electro-Communications, Tokyo, Japan, in 1984 and 1986 respectively. He also received the D.E. degree in mathematical engineering and information physics from the University of Tokyo, Japan, in 2001. In 1986, he joined NTT (Nippon Telegraph and Telephone Corp.). Since 1988, he has been with the University of Electro-Communications and is currently an Associate Professor in the Department of Information and Communication Engineering at the same University. His research interests are in the areas of wave Signal Processing, image and Acoustic Signal Processing, and electronic measurements.

Automatic Film Line Scratch Removal System based on Spatial Information ★
Kyungtai Kim (Konkuk University, Korea); Eun Yi Kim (Konkuk University, Korea)
Film restoration is to detect the location and extent of defected regions from a given movie film, and if present, to reconstruct the lost information of each regions. In the recent year, Film restoration has gained increasing attention by many researchers, to support multimedia service of high quality. In general, an old film is degraded by dust, scratch, flick, and so on. Among these, the most frequently degradation is the scratch. The old film includes many kinds of scratches. Therefore, many systems have been developed for automatic scratch restoration. However, before method has some problem. Because, it fails to detect not-alone scratches or secondary scratches in detection step. Moreover it produces many false-alarms and dismissal in the sequence with complex background. To resolve these limitations, this paper presents a scratch restoration method that automatically detects all kinds of scratches from frames in old films. In this paper, an automatic film line scratch removal system is developed that can deal with all kinds of scratches. For this we use the spatial information of scratches: the scratch in old films has lower or higher brightness than neighboring pixels in its vicinity and it usually appears as a vertically long thin line. Our system consists of two major modules: detection using characteristics of a scratch and removal. First step is to detect the scratches included in a frame using a neural network-based texture classifier and a morphology-based shape filter with multiple structuring elements. The input image is first divided into scratch regions and non-scratch regions by a NN-based texture classifier, and then some over-detected regions are filtering by morphology-based shape filter with new structuring elements. To remove the degraded regions after scratch detection, we use bilinear interpolation method as the spatial restoration algorithm. The bilinear interpolation is a method to restore damaged parts of an image by filling in the region with surrounding information. To assess the validity of the proposed method, the experiments have been performed on several old films and
Recovery Chromatic Information of Images Spoiled by Near Infrared Ray

Jinlong Lin (Peking University, P.R. China); Chao Zhang (Peking University, P.R. China)

COMS image sensors are widely used in digital image and video products, however, the nearly equal and strong response to near infrared ray of sensor’s three channels causes the chromatic difference between the captured images and their original scenes. In this paper, we present a new method to solve the problem. After finding the reference pixels whose \((R+G+B)/\min(R, G, B)\) are greater than a threshold, we estimate the global rate of grey value by near infrared ray to brightness of pixels according to the surface reflection mode and the response characteristics of three channels. The final images are generated by subtracting the estimated infrared ray induced grey value from \(R, G, B\) channels for all pixels. The experimental result shows that the proposed algorithms can eliminate the effects of infrared ray and recovery chromatic information of original scenes without optical high pass filter.

Novel CAT Wavelets-based Image Coding System

Rong-Jian Chen (National United University, Taiwan); Cheng-Fang Tai (National United University, Taiwan); Jui-Lin Lai (National United University, Taiwan)

The goal of this paper is to present a novel image compression method using the orthogonal cellular automata transform (CAT) bases. The applications of the novel image compression method can be consumer applications such as multimedia devices, client/server communication, military/surveillance, medical imagery, digital libraries/archives, and E-commerce. The first aim of this paper is to investigate the CA evolution rules, which will result in the orthogonal CAT bases. The second aim of this paper is to establish the hierarchical CAT-decomposition scheme to yield CAT wavelets. Finally, the CAT-based image coding system using CAT wavelets is proposed. The proposed image coding system consists of the hierarchical CAT-decomposition scheme, the coding schemes, and the hierarchical inverse CAT (ICAT)-reconstruction scheme. In the compression site, we select a set of suitable CA initial state and CA generating rule to make the orthogonal CA bases generator to yield a set of orthogonal CA bases for hierarchical CAT decomposition. After hierarchical CAT decomposition, the original image becomes to the CAT wavelets. We then use tree-structure encoding scheme to produce the ordered bit stream of CAT wavelets for transmission or storage. In the decompression site, firstly, the CA initial state and CA generating rule are separated from the bit stream for generating a set of orthogonal CA bases for hierarchical ICAT reconstruction. Next, we use tree-structure decoding scheme to produce the reconstructed CAT wavelets, and then using the hierarchical ICAT reconstruction scheme to obtain the reconstructed image. Simulation results for some gray images show that the performance of the proposed CAT wavelets coding system superior to the JPEG standard and achieve some of that of traditional SPIHT. Reasons of the proposed CAT wavelets coding system being superior to the JPEG standard are: (1) Orthogonal CA bases, which are evaluated from CA, have excellent properties of energy compaction. (2) The CAT/ICAT process only consists of integer additions and subtractions, whereas the DCT/IDCT operation consists of floating-point multiplications, additions, and subtractions. (3) The CAT wavelets are hierarchical in that they are suitable for SPIHT encoding and decoding. (4) The proposed coding method accomplishes completely embedded coding and progressive transmission. The performance of the proposed CAT wavelets coding system is so impressive that it can be a suitable candidate for inclusion into standardization in future image compression systems.

Rong-Jian Chen (M2002, SM2004 IEEE) received the B.S., M.S., and Ph.D. degrees in electronic engineering from the National Taiwan University of Science and Technology, Taipei, Taiwan, in 1987, 1991, and 1995, respectively. He joined the faculty of the National Lien-Ho Institute of Technology in August 1995, where he was the Chair of the Department of Electronic Engineering during the 1999-2001 academic years. At present, he is Associate Professor of the Department of Electronic Engineering, National United University. His research interests include digital image/video signal processing, image/video coding and encryption, pattern recognition, neural networks, VLSI architecture design and implementation for image/video coding and encryption, VLSI architecture design and implementation for neural network systems, and globalization leadership. He is a member of Nano-electronics and Giga-scale System Technical Committee, IEEE CAS Society. He serves as Co-Editor of Globalization Leadership Column at IEEE Circuits and Devices Magazine (per invitation from EIC Dr. Ron Waynant), and serves on the Editorial Board of IEEE Circuits and Systems Magazine (per invitation from EIC Prof. Maciej Ogorzalek). He also serves as Associate Editor of IEEE Trans. on Circuits and Systems, Part 1 from Dec. 15, 2005. He also serves on the Technical Program Committee of 2004 IEEE ICECS (International Conference on Electronics, Circuits and Systems) at Tel-Aviv, Israel in Dec. 2004. He served on the Organizing Committee of 2003 IEEE International Symposium on Nanoelectronics and Giga-scale Systems, Miaoli, Taiwan, Oct. 2003, which was technically co-sponsored by IEEE Circuits and Systems Society and IEEE Electron Devices Society. Prof. Chen also serves on...
the Organizing Committee of 2004 International Workshop on Efficient and Competitive Professors to be held in Hsin-Chu, Taiwan, Dec. 2004. He serves as Review Committee Member for 2005 IEEE ISCAS Symposium, in Kobe, Japan, Review Committee Member for 2006 IEEE ISCAS Symposium, in Kos Island, Greece, and the Technical Program Committee of 2006 IEEE ICECS at Nice, French in Dec. 2006. He also serves as the Technical Program Committee of 2007 IEEE ICECS.

Gaming Platforms & Systems

Room: Trinity 1-3  Chair: Corey Carbonara (Baylor University, USA)

Fusion Technique for User Identification using camera, microphone and ultrasonic sensor in the intelligent service robot

Kyu-Dae Ban (University of Science and Technology, Korea); Keun-Chang Kwak (ETRI, Korea); Hosub Yoon (ETRI, Korea); Yun-Koo Chung (ETRI, Korea)

In this paper, the fusion technique inclusive of face information and speaker information for the intelligent service robot is proposed. We proposed score based fusion, this evidence scores were obtained through a face recognition and speaker recognition. For the face recognition, we use appearance-based face recognition techniques, eigenface method, which is based on PCA approach. And for the speaker recognition, we utilize GMM approach which uses a MFCC as feature vector. The hybrid fusion techniques provided better performance than using the face information and speaker information experts alone. Additionally, we propose the method to obtain human height information. Human height is obtained through utilizing the cheap ultrasonic sensor and camera parameter such as tilt and field of view. The experimental results obtained through the research robot platform called WEVER developed in ETRI, Korea.

An Ad Hoc Wireless Remote Sensor Networking System using the TI MSP-430

Michael Helm (Florida State University, USA); Anthony Presley (Resolution Systems, USA); Erik van Gilder (Resolution Systems, USA)

A networked system of wireless remote data sensors is described. The system includes multiple standalone remote sensor nodes and nodes to control and coordinate data collection. Capability for ad hoc network expansion is included. Battery life considerations are addressed by using the low power TI MSP-430 to handle control and data transfer and by using a power control algorithm to minimize power consumption of the wireless RF subsystem. The paper includes results from empirical testing of the prototype system. Application areas include a wide variety of data collection and security needs.

Dr. Michael Helm has been involved in the development of electronic systems for nearly 30 years. He has worked on microwave communications systems, microprocessor based embedded systems, automation systems, and test systems. He worked for Texas Instruments for 19 years primarily in the Consumer Products Division. After recently completing a PhD, he left his home in Texas and began teaching Computer Engineering courses at Florida State University in Panama City, Florida. He continues to be involved in product development work, currently in the area of wireless sensor systems using the CC2420 and MSP430.

Security & Digital Rights Management 2

Room: Trinity 4

Fingerprinting Algorithm for Tracing Colluders

Jung Soo Lee (ETRI, Korea); Ki-Song Yoon (ETRI, Korea); Young-Ho Suh (ETRI, Korea)

In this paper, we propose a fingerprinting algorithm that is robust to collusion attacks and can trace colluders. To do this, we design the structure of the fingerprinting code and fingerprinting technique based on a watermarking scheme that uses a 'base watermark set', which composes the information to be embedded. In addition, we propose the structure that colluder trace system can use various fingerprinting algorithm, which is needed to insert many 'base watermarks' into the contents. We will check the quality of the content after inserting fingerprinting according to the proposed structure. In the experiment, we mainly focused on robustness against collusion attacks and showed that our algorithm was robust to various types of collusion attacks.

Jung-Soo Lee received his Ph.D. degree in Electronic Engineering from Hanyang University, Seoul Korea in 2005. Currently, he is a senior member of Electronics and Telecommunications Research Institute(ETRI). His research interests are Digital Watermarking, Fingerprinting, Image Processing and Digital Rights Management.

Preventing Method Illegal or Harmful Contents from Spreading in P2P network
In this paper we propose a method that prevents illegal or harmful files from spreading in P2P network. We apply the idea of Honeypot to P2P network. We build fake P2P service farms and monitor and trace users who spread or gain illegal or harmful files in P2P network. If we can apply this system widely, we can expect illegal or harmful files to be wiped out in P2P network. P2P network is a network technology that has made a great stride from early 1990. It began with Napster in USA and Soribada in South Korea. The existing network model (client and server model) needs a central control of server but P2P network connects each peer directly. So it can decentralize network traffic and utilize network efficiently and spread information very quickly. But users started to distribute illegal mp3 or harmful videos in P2P network because there are no surveillances. The illegal/harmful contents problem became an issue and many prevention measures are proposed[1]. The prevention measures are divided two. The one is an approach based on social engineering and the other is an approach based on technology. Until now an approach based on social engineering has been adopted for most cases. The approaches based on social engineering prohibit a certain P2P service by law or trace and indict some peoples who spread illegal files. The approaches based on technology are as follow. The first one is that every user who uses P2P network installs agent software in his computer. The agent software monitors the P2P traffics and prevents the P2P traffics if user uploads or downloads illegal files. This approach must install and manage agent software on every personal computers but it is almost impossible. The second one is that contents monitoring system is installed on network border and the system monitors all P2P traffic. This approach must install contents monitoring systems on every network border and too much cost is needed and the development of contents monitoring system that can detect all P2P traffic in real time is very difficult. So in this paper, we propose the new technical approach to prevent illegal and harmful contents from spreading in P2P network. It adopts the idea of Honeypot and monitor and trace illegal users. The proposed approach can gather forensic evidence that a user spreads illegal files. In this paper we adopt the idea of honeypot and design a method to trace users who spread illegal or harmful contents in P2P network and store forensic evidences. To do this, we operate several P2P service on bait P2P server farm. To gather illegal/harmful contents list from these bait P2P server, we need to analyze each P2P service protocol and hook and reassemble the P2P packets and extract file lists. And then, we store these file list in central DB. From this information we can figure out the trend of distribution of illegal/harmful contents. Each file on file list can be classified automatically to illegal/harmful or not using machine learning. In chapter 4, we show these machine learning algorithms. To gather files in file list, we use the P2P client function. So our bait P2P function can execute server function and client function. If our classification module judges the file illegal/harmful, we store the forensic evidences that are source/target IP, request time, target file and user etc. First of all we install P2P server farm to allure the users who spread illegal/harmful files. In honeypot, the honeypot server provides virtual services but in P2P honeypot, the bait P2P server provides real P2P services. The bait P2P server farm provides real open P2P service like eDonkey, emule, Pruna(in Korea). And the bait P2P server farm can have several kinds of servers. Second, there is a packet hooking and data gathering module. We analyze P2P protocols and we know the meaning of the each P2P packet data field. In the search request of client, we collect keyword and file list that the user has and store those information in function DB. A data gathering module classifies and stores all these information from all P2P servers. Third Data Managing and Analyzing module. Most of search engines store their URL information efficiently in their optimal data structure and provide URL ranks or hit information. Like this, P2P honeypot can calculate the trend of illegal/harmful files in the P2P network. If a certain file has spread explosively in short time, the file may be illegal or harmful file (ex. Hottest movie file or sex video of famous actress) Forth, Automatic illegal/harmful file classification module. In this part we use two algorithms to inspect a file. From search keywords or file names, we can infer a content of the file. So we apply text classification algorithm to those information. And the most harmful contents are videos. So we download these videos using client function and apply video classification algorithm to classify harmful or not. Fifth Tracing module with hash code. With Automatic illegal/harmful file classification module, we can not provide 100% accuracy due to weakness of machine learning algorithms. But in some case we need 100% accuracy. To support these cases, we can trace a certain file using hash code. If we need to trace the hottest movie file or sex file of famous actress, we can make the hash code of those file and upload those hash code to P2P hooking and examine module. This module can examine all P2P file with hash code using P2P client function. The hash code is made by file name, file size and file header. Until now, we show five parts of P2P honeypot. With these modules, we can monitor and trace the illegal/harmful files in P2P network.

HoGyun Lee received the MD degree in Computer Engineering from the Kungpook National University. He is currently a research engineer in privacy protection research team of Electronics and Telecommunications Research Institute. His current research interests are the DDoS detection, Harmful Contents Prevention, and Pattern Recognition.
As the home devices have various functions and have improved computing power and networking ability, the importance of home device authentication is increasing for improving of home network users’ security. In using home network service, user authentication and authorization technology are applied to home network services only for authorized persons to use the home network services. But it has some problems: the leakage of user authentication information by user’s mistake, usage of guessable authentication information, and finding of new vulnerability about existing authentication method. So it is necessary that home network service user can be served the secure home network service by only using credible device. This means that home device authentication besides user authentication and authorization is essential to the secure home network service. Also, the unauthorized accessing possibility for our home network is very high by the device included in neighbor home network because of the home network characteristic; various wired/wireless network devices is used in the home network. This is an additional reason about the necessity of device authentication. Finally, we think that the secure relationship among home network devices is very important factor because home network service evolves into more convenient one; user’s role in receiving home network service is minimized and the number of service served by cooperation among home devices increases. This paper proposes home device authentication mechanism using PKI. It covers intra-home device authentication and inter-home device authentication. We consider public key infrastructure not using personal CA but using public CA We propose home device certificate profile for home device authentication and describe home device registration and certificate issuing method. Also, we describe home device authentication method and its relationship with home device certificate fields. Our home device authentication framework is able to protect home network users from malice accessor of home network. And it provide user convenience

Fuming Wu is currently a senior member of the IEEE and a professional member of the ACM. He has been filed under Patent Cooperation Treaty in the International Bureau of World Intellectual Property Organization. He is an author or co-author of a number of research papers published in refereed journals and international conference proceedings. Fuming Wu is now a Co-PI of the NSF S-STEM ME-Best Program with awarded grant $500K. Fuming Wu is now an assistant professor of computer science and a senior member, scientific staff, in a systems engineering group in its Richardson Lab (Richardson, Texas), Nortel Networks Corp. Fuming Wu’s research interests cover theoretical aspects of computer science, dynamic modeling of information systems using high-level timed Petri nets, GMPLS controlled optical networks, mobile ad hoc and sensor networks, fixed and wireless Internet networking and applications, and emerging technologies in the fields of networking, information security, and bioinformatics. He is an inventor with 17 (pending) patents filed in the European Patent Office, the US Patent & Trademark Office and the State Intellectual Property Office of the People’s Republic of China. Several of his patent applications have been filed under Patent Cooperation Treaty in the International Bureau of World Intellectual Property Organization. He is an author or co-author of a number of research papers published in refereed journals and international conference proceedings. Fuming Wu is now a Co-PI of the NSF S-STEM ME-Best Program with awarded grant $500K. Fuming Wu is currently a senior member of the IEEE and a professional member of the ACM. He has served as a member of technical program committee of numerous international conferences and a peer reviewer of Journal of Supercomputing, Journal of Research and Practice in Information Technology (JRPIT), the 2004 International Conference on Communications in Computing (IEEE ICC’04), and the 2007 Australasian Information Security Workshop – Privacy Enhancing Technologies (AISW’07).

10:45 AM - 12:00 PM

Presence Technology with Its Security and Privacy Implications ⭐
Fuming Wu (Texas A&M International University, USA)

With its broadening scope, presence technology is gaining significant interest and becoming an indispensable component in nowadays telecommunications industry as well as in the vertical market. Presence-based services are driving new business opportunities and changing every aspect our daily life. Meanwhile, with the sensitive nature of rich presence information, security and privacy issues are among the greatest concerns to the end-users of presence-based services. In the present paper, we briefly introduce the presence technology, the evolution of the presence notion and the fundamental model of presence service. We also give an overview of presence-based services together with example implementations of presence-enhanced directory services in enterprise environment. Understanding the presence technology and presence-based services together with their design issues is an important prerequisite to discuss security and privacy implications of presence-based services.

Fuming Wu received his PhD degree in Mathematics and Theoretical Computer Science from University of Haifa, Israel, in 1998. He holds MS degrees in Mathematics and in Networking and Telecommunications, and a BS in Mathematics and Mathematics Education. Fuming Wu is now an assistant professor of computer science and a member of graduate faculty in Full/Doctoral level at Texas A&M International University (Laredo, TX). He had more than five years’ industrial experience: He was a research scientist in the Research & Innovation Center (Plano, TX), Alcatel USA Resources, and a senior member, scientific staff, in a systems engineering group in its Richardson Lab (Richardson, Texas), Nortel Networks Corp. Fuming Wu’s research interests cover theoretical aspects of computer science, dynamic modeling of information systems using high-level timed Petri nets, GMPLS controlled optical networks, mobile ad hoc and sensor networks, fixed and wireless Internet networking and applications, and emerging technologies in the fields of networking, information security, and bioinformatics. He is an inventor with 17 (pending) patents filed in the European Patent Office, the US Patent & Trademark Office and the State Intellectual Property Office of the People’s Republic of China. Several of his patent applications have been filed under Patent Cooperation Treaty in the International Bureau of World Intellectual Property Organization. He is an author or co-author of a number of research papers published in refereed journals and international conference proceedings. Fuming Wu is now a Co-PI of the NSF S-STEM ME-Best Program with awarded grant $500K. Fuming Wu is currently a senior member of the IEEE and a professional member of the ACM. He has served as a member of technical program committee of numerous international conferences and a peer reviewer of Journal of Supercomputing, Journal of Research and Practice in Information Technology (JRPIT), the 2004 International Conference on Communications in Computing (IEEE ICC’04), and the 2007 Australasian Information Security Workshop – Privacy Enhancing Technologies (AISW’07).
**Audio Video Technology 6 ★**

**Room: Trinity 6-8**

**Correction of over-exposed images captured by cell-phone cameras ★**

Qolamreza Razlighi (University of Texas at Dallas, USA); Nasser Kehtarnavaz (University of Texas at Dallas, USA)

Due to the limited dynamic range of commonly used cell-phone camera sensors, the entire or part of an auto-exposure (AE) image may appear over-exposed. In this paper, we present an over-exposure correction technique which utilizes one low-exposure (LE) image in addition to the AE image. It consists of registration, fusion, and blending components. In all these components, computational efficiency is taken into consideration to allow deployment on cell-phone processors having limited processing power. Registration is achieved by using a one-dimensional grid correlation approach. A linear weighted summation is adopted to achieve fusion. It is shown that to obtain a high contrast fused image, it is necessary to perform subarea or block-based fusing of the two images. A mechanism based on image entropy and histogram is developed to adaptively alter the required weighting factors for appropriately fusing the subareas of the AE and LE images. The optimum weighting factors are obtained by formulating an optimization problem which is made computationally efficient via histogram modeling. The fusion process creates blocking artifacts at the borders of subareas. A Gaussian function is then used to remove such artifacts by smoothly blending the neighboring subareas. The results obtained show that our solution is more effective than the existing and comparable overexposure correction techniques. A sample overexposure corrected image is shown in Figure 1 together with the AE and LE images. (for figure 1 please refer to the PDF manuscript)

Nasser Kehtarnavaz (S’82-M’86-SM’92) received the PhD degree from Rice University, Houston, TX, in 1987. He is currently a professor with the Department of Electrical Engineering and Director of the Signal and Image Processing Lab at the University of Texas at Dallas. His research interests include digital signal and image processing, real-time image processing, and pattern recognition. He has authored or co-authored five books and numerous articles as related to these areas. Dr. Kehtarnavaz is currently serving as Chair of Dallas Chapter of IEEE Signal Processing Society.

**Towards Practical Print-from-video: Improved Pattern Filtering ★**

Qingzhong Peng (University of Texas at Dallas, USA); Nasser Kehtarnavaz (University of Texas at Dallas, USA)

ABSTRACT Print-from-video can be achieved by using super-resolution (SR) techniques [1]. These techniques involve combining information from multiple low resolution images to generate and print a high resolution image. Among the existing SR techniques, the most suitable technique for deployment on portable consumer electronics products having limited processing and memory resources is the shift-and-add method [2]. However, this technique is known to generate artifacts when applied to real video sequences. This paper introduces a number of improvements made to the shift-and-add technique to reduce the level of artifacts so that it can be deployed on portable camera devices. These improvements, named improved pattern filtering, consist of the three computationally efficient steps of frame alignment, frame selection, and frame fusion. In these steps, motion vector field and DCT coefficient information are derived from an MPEG codec and utilized to improve the quality of printed images. The effectiveness of the developed solution is demonstrated by showing both quantitative and qualitative comparison results. A sample reconstructed image is shown in Figure 1, where the improved outcome appears on the right. <Please refer to submitted abstract in PDF form for Figure 1> Fig. 1. A sample super-resolution reconstructed image from a handshake blurred video sequence using (a) the shift-and-add algorithm, and (b) the improved pattern filtering algorithm. REFERENCES 1. S. Park, M. Park and M. Kang, “Super-resolution image reconstruction: a technical overview,” IEEE SP Mag, vol. 20, pp. 21-36, 2003. 2. M. Elad and Y. Hel-Or, “A fast super-resolution reconstruction algorithm for pure translational motion and common space invariant blur,” IEEE Trans. Image Processing 10, pp. 1187-1193, Aug. 2001.

Qingzhong Peng received the Bachelor of Science degree in Electrical Engineering in 2005. He is currently pursuing the Master of Science degree in Electrical Engineering at the University of Texas at Dallas as a research assistant in the Signal and Image Processing Laboratory. His research interests include super-resolution image reconstruction and real-time image processing for digital camera applications.

**Image restoration preprocessing for low-light auto-focusing in digital cameras ★**

Mark Gamadia (University of Texas at Dallas, USA); Nasser Kehtarnavaz (University of Texas at Dallas, USA)

The performance of passive auto-focus (AF) systems suffers when operating under low-lighting conditions due to the flatness of the sharpness function. In this paper, an adaptive noise reduction method prior to AF sharpness...
filter is introduced for the purpose of elevating the suppressed peak in the sharpness function. Experimental results show that the use of adaptive noise reduction preprocessing enables focusing at lower lux levels as compared to non-adaptive noise reduction preprocessing.

Mark Gamadia (S'06) received the BS in electrical engineering from the University of Texas at Dallas (UTD), Richardson, TX, in 2003, the MS in electrical engineering from UTD in 2004. He is currently a PhD candidate and Erik Jonsson Fellow in the Department of Electrical Engineering at UTD. His research interests include real-time video/image processing with applications in consumer electronics such as digital cameras. He has co-authored the book Real-Time Image and Video Processing: From Research to Reality and several papers regarding real-time implementation of image processing algorithms on digital camera platforms.

**Comparison of power consumption for motion compensation and deblocking filters in high definition video coding**

Jianfeng Ren (University of Texas at Dallas, USA); Nasser Kehtarnavaz (University of Texas at Dallas, USA)

Abstract — From a computational perspective, it is known that motion compensation and deblocking constitute the two most time consuming components in the video decoding standard. This paper presents a comparison study of the commonly used motion compensation and deblocking filters in terms of the number and type of instructions to indirectly analyze the level of power consumption. Video quality is also taken into consideration as part of this comparison. The comparison results show that the motion compensation and deblocking filters adopted by the H.264 standard provide the highest video quality at the lowest power consumption level for low bit rate video coding applications. Index Terms — Video coding power consumption comparison, deblocking filter, motion compensation, high-definition video.

**Algorithmic Optimization for Low Power H.264 video decoding on portable devices**

Jianfeng Ren (University of Texas at Dallas, USA); Nasser Kehtarnavaz (University of Texas at Dallas, USA)

Abstract — Deblocking filters are used to reduce or eliminate blocking artifacts that appear when H.264/AVC is utilized for low bit rate video coding applications such as video conferencing. Considering that the H.264/AVC deblocking filter constitutes about one-third of the decoding time, in this paper, we present two optimization steps to reduce the computational complexity of this filter. These steps involve using the mode information of decoded macroblocks and the similarity among adjacent blocks. Experimental results show that these optimization steps significantly reduce the computational complexity with tolerable loss in video quality. Index Terms — Optimization of H.264/AVC deblocking filter, mode selection in video deblocking, block similarity in video deblocking.

**Consumer Networks 1**

Room: Trinity 1-3

**A study on the method of providing family finder on TV services in the open service platform**

HyunKyung Yoo (Electronics and Telecommunications Research Institute, Korea)

Until now, PSTN, internet, mobile telecom network and broadcasting network providers have provided the services with their own network resources. According to the innovative development of digital information technology, our society is evolving to the NGN(Next Generation Network) that combines the diverse telecom and broadcasting network into a converged network. For various NGN services, it needs to open network resources to 3rd party service providers. And then, service providers enable to develop and provide the more converged services with open API technology. Parlay X gateway using open API has service functions of third party call, call notification, short messaging, multimedia messaging, payment, terminal status and location, call handling, audio call, multimedia conference, address list management, presence, message broadcast, multimedia multicast control and Geocoding. Service developers can generate services easily by calling these service functions. Like this, open API can present a unified face to service developers under which can be hidden the details of network protocols that are not of interest to developers. In this paper, we described the architecture and functions of open service platform that is composed of Parlay X application, Parlay/Parlay X gateway, GMLC/MPC, addressing server and mobile terminal. And we defined open APIs that supported the location address of mobile subscriber and provided the concept and control flow of the family finder on TV services by using open API. Family finder on TV services support the location address based on the position of mobile subscriber to requestor on watching TV. They are composed of the member positioning service and the member safety service. The former provides the location address of mobile subscriber and provided open API technology.

I work for ETRI. ETRI is Information and Telecommunications research institute for Government. I'm working in

http://edas.info/showProgram.php?_qf__showProgram=&c=5136&form...bio%5D=1&program_view=everybody&action=Show+conference+program Page 35 of 81
Converged Service Research Team.

**Design and Implementation of Software architecture for Open Home network Framework ★**

HoonKi Lee (Electronics & Telecommunications Research Institute, Korea)

Through Home Network market's activation today, application service of various forms is offered. Finally, diversification of service need to offer environment that can provide service that want always to user. We need the framework to accept such environment change actively. This paper explains Open Home network Framework (OHF)'s software structure that can provide home network base application service of hardware independent. And describe about design and implementation of reference Home Server system based on OHF. Now, Framework of Home network exists in various forms with OSGI, IGRS, THAI, and ECHONET. Also, this framework offers structure of each diagnostic form. The Open Home Network Framework (OHF) is an infrastructure which provides application programs and network interoperability, quality assurance, and secure environment for various services based on user preferences in the next generation home network. Several framework structures that offer in home network environment are offered for single service offer or service management purpose. However, OHF wishes to offer communication, broadcasting, and basis structure for multiplex service offer that consider aspect control in several frameworks. Frameworks that have each diagnostic structure include basically in present OHF or accommodation wishes to offer possible structure to OHF. OHF framework wishes to offer Home network framework that can trust considering all user side and service provider side. OHF's basis structure is consisted of three Layer and Service API. These structure offers comprehensive function provide about broadcasting, communication, and control service that is offered in Home network. OHF is doing by purpose that supply multiplex or service of form that download is possible to various service user minimizing service interworking. Each Layer's function provides Home network service, and accommodates basis function such as interworking between services through reliable control of service. Also, offer availability about Legacy service is consisted of open home network structure that accommodate. In this paper described home network base application service of various form by explain and do actuality design and Implementation home server and reference style system through this about Open Home network Framework's basis structure. (Include OHF's architectures structure and design through reference home server and implementation through Full paper)

**Efficient Traffic Prediction Algorithm of Multimedia Traffic for Scheduling the Wireless Network Resource ★**

Kang Yong Lee (ETRI, Korea); Cho Kee Seong (Access Mediator Research Team, ETRI, Korea)

Real-time Variable Bit Rate (VBR) video traffic generated from multimedia applications, such as MPEG-coded video stream, is expected to be large portions of the traffic in future wireline/wireless networks. The natures of VBR traffic and its Quality of Service (QoS) constraints increase a number of challenges on the network resource managements and operational utilizations. Thus, accurate traffic prediction based dynamic bandwidth allocation and scheduling can significantly improve network performance substantially while satisfying the QoS requirements. In this paper, we propose a novel algorithm for predicting MPEG-coded real-time VBR video traffic. Based on statistical property of traffic stream, our algorithm performs a prediction of the next frame size for the I-, P- and B-frames. Simulation results using real-world MPEG-4 coded video traces show that the proposed algorithm achieves much better performance than the LMS and NN method. In addition, the applicability of our prediction algorithm to IEEE 802.11e Wireless LAN is discussed.

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**Security & Digital Rights Management 3 ★**

Room: Trinity 4

**Design and Implementation of Mobile User Agreement Service ★**

Seung-Hyun Kim (Electronics and Telecommunications Research Institute, Korea); Han-Gyu Ko (UST, Korea); Seung-Hun Jin (ETRI, Korea)

Recent IT technologies turn to the direction to provide personalized service which reflects a person's favor, not just deliver uniform services and information. Such as Web 2.0, it is getting more popular for users to participate in creating contents and sharing their information. Shared information includes not only ordinary information but also user’s private information which is scattered on several sites in the Internet. However there is no suitable approach to manage user’s private information systemically from a legal, technical point of view. Because service providers recognize user’s private information as their belonging, it may bring a serious threat of privacy infringement. As sharing for user’s private information is more usual, a specific system is required to protect user’s privacy.
Traditional researches propose various methods which make a user to set policies to manage his/her private information. A representative method among them is XACML. XACML sets a policy at PAP (Policy Administration Point). If a request is received, XACML evaluates an applicable policy at PDP (Policy Decision Point) and performs access control at PEP (Policy Enforcement Point). However traditional methods have a weakness that a user has to set all policies by himself/herself or has to rely on the preset policies created by an administrator. The more service providers and exposed private information, the less methods to apply. Because sharing of private information largely depends on the specific circumstance and purpose, it is not flexible to decide sharing of private information with preset policies. We need a rational process that sets minimum policies and handles a task with the real-time user agreement for sharing of private information. To handle a task with the real-time user agreement, limitation of application area such as specific area or fixed target is not allowed. Therefore we propose an architecture to query a real-time user agreement in the mobile environment. The proposed architecture consists of 5 entities named AP (Attribute Provider), IS (Interaction Server), MC (Mobile Client), AC (Attribute Consumer) and CS (Certificate Server). The architecture provides two main services; a certificate transport service and a real-time user agreement service. The Certificate transport service moves a user’s certificate and a private key located in the user’s PC to MC with PKCS#12[12] technology. The user’s certificate is used for providing high level of security. And the real-time user agreement service starts with sending a SMS message which contains Callback URL from IS to MC. MC establishes secure communication channel between MS and IS, and then sends a user’s agreement signed with the user’s certificate to IS. With the conjunction of wired environment, the proposed architecture is able to provide two-factor authentication. Also it is possible to create a new business model such as mobile auction. We have implemented the proposed architecture at the SKT network which is the largest 3G mobile network in Republic of Korea. And we have a plan to commercialize the proposal.

**a Unified User Consent Acquisition and Delivery Mechanism for Multi-Source User Data Integrated Service**

YoungSeob Cho (Electronics and Telecommunications Research Institute, Korea)

Internet service providers have usually collected and maintained user data necessary for their services. Recently, many SP(Service Provider)s supply users with integrated services which combine existent user data of other service providers. When UdP(User Data Provider) provides SP with user data, it should acquire user consent to preserve user privacy and to avoid future responsibility. However, UdP has not direct session with user, so it is very difficult that the UdP acquires user consent directly from the user. In addition, if user may give its consent base on individual UdP, this may be inconvenient for user. In this paper, we propose a unified user consent acquisition and delivery mechanism for multi-source user data integrated service. We introduce DA(Delegation Authority) for user consent acquisition and delivery. DA acquires user consent to UdP’s data providing from user and generates an ELA(Electronic Letter of Authorization) from user consent information, and sends it to SP. SP sends the ELA with user data request to UdPs, which use the ELA for deciding whether to provide user data. We design ELA Scheme, message protocols and other components such as bindings, metadata and identifier. The proposed mechanism enables user to control explicitly its own data flow and to give its consent to all SP service-related UdPs only for one interaction.

**Identity Roaming Mechanism for Multiple Trust Domains using SAML v2.0 and Single Federation Bridge**

Sang Rae CHO (ETRI (Electronics and Telecommunications Research Institute), Korea); YoungSeob Cho (Electronics and Telecommunications Research Institute, Korea); Seung-Hun Jin (ETRI, Korea)

The Web has been evolutionally evolved for the last decade. However, the fundamental technology that processes the subscription and authentication of an ordinary user has not been improved a bit for last two decades. When we visit a website to use some service, we have to provide our identity information to join the web site. After that, we have to authenticate ourselves to the web site every time we visit that website. In order to get rid of this inconvenience, the federation identity has been introduced to connect distributed account information for Single-Sign-On (SSO) service. SSO enable a website to share its authenticated information with another website using federated identity link. The identity provider (IDP) is the term that describes the entity that authenticates a user and distributes the fact of authentication to another website. The relying party is called a Service Provider (SP) that consumes the authentication statement that is issued from an IDP. If a SP wants to assure an authentication statement from an IDP, there has to be a trust relationship between two parties. We call this trust domain Circle of Trust (CoT) that consists of one or more IDP and several SPs. The concept of CoT has a problem if the number of CoT grows exponentially. In this situation a user has to authenticate himself to a CoT every time he visits. We have the same problem in our hands. Therefore, we have concluded that identity roaming technology is the only solution to overcome this problem in the federated identity network. In this paper, we have proposed noble method of the identity roaming to enable a user to sign up for one web site and extend the connectivity service in a location that is different from the home location where the user was registered and authenticated. The identity roaming occurs when a subscriber of one web site uses the facilities of another web site that is a member of a
A Privacy-Enhanced Adult Certification Method for Multimedia Contents on Mobile RFID Environments

Namje Park (ETRI, Korea); Younusu Kim (ETRI, Korea); Dongho Won (SKKU, Korea)

Abstract. Recently, RFID (Radio Frequency IDentification) technology is practically applied to a number of logistics processes as well as asset management, and RFID is also expected to be permeated in our daily life with the name of ‘Ubiquitous Computing’ or ‘Ubiquitous Network’ within the near future. The R&D groups in global now have paid attention to integrate RFID with mobile devices as well as to associate with the existing mobile telecommunication network. Such a converged technology and services would lead to make new markets and research challenges. However, the privacy violation on tagged products has become stumbling block. We propose a privacy-enhanced method of certifying adults on WIPI (Wireless Internet Platform for Interoperability) in mobile RFID environments. This Paper is for checking a user whether he/she is an adult or not, when a user would like to read some adult-related data using mobile devices playing a role of tag readers on mobile RFID environments. Instead of current adult-certification method using one’s own mobile devices, an adult certification in this paper uses any mobile devices and provides user’s anonymity. 1 Introduction We propose a privacy-enhanced method of certifying adults on WIPI (Wireless Internet Platform for Interoperability) in mobile RFID environments. This Paper is for checking a user whether he/she is an adult or not, when a user would like to read some adult-related data using mobile devices playing a role of tag readers on mobile RFID environments. Instead of current adult-certification method using one’s own mobile devices, an adult certification in this paper uses any mobile devices and provides user’s anonymity. 2 Some Mobile RFID-oriented Privacy Threats Security vulnerability of the mobile RFID are the infringement of owners’ privacy and the physical attack in cyber space. Typical vulnerabilities are as follows: - Privacy Threats Individual privacy is very likely to be infringed due to the approval of unlimited ac-cess to a RFID tag owned by a person. So, it is necessary to allow the access to in-formation only for who needs to get it in a given application but to bock it for those who have no authority. An individual RFID tag also may become means to track and locate its owner. The infringement of privacy in the internet world results from the collection, storage, and use of customers by companies, but it has grown more serious in the mobile RFID world in that any one with RFID reader can read any in-formation on any one who keeps a tag attached object. - Possible hacking of mobile RFID applications It is possible to hack tags, preventing to use normal tags or to get incorrect infor-mation from them with a tag information alteration or a tag kill function. - Illegal collection of information One may hide a system in commodity or object for remote communication to wire-tap, track, catch information, or profile. - Security vulnerability from jamming, replay attack, covert reading, etc. 3 Key Technology and Solution This technology aims at RFID application services like authentication of tag, reader, and owner, privacy protection, and non-trackable payment system where more strict security is needed. 1) Approach of Platform Level This technology for information portal service security in offering various mobile RFID applications consists of application portal gateway, information service server, terminal security application, payment server, and privacy protection server and provides a combined environment to build a mobile RFID security application service easily. 2) Approach of Protocol Level - It assists Write and Kill passwords provided by EPC Class1 Gen2 for mobile RFID tag/reader and uses a recording technology preventing the tag tracking. - It employs information protection technology solving the security vulnerability in mobile RFID terminals that accept WIPI as middleware in the mobile RFID reader/application part and provides end-to-end security solutions from the RFID reader to its applications through WIPI based mobile RFID terminal secu-rity/code treatment modules. 3) Approach of Privacy Level This technology is intended to solve the infringement of privacy, or random acquisi-tion of personal information by those with RFID reader from those with RFID attached objects in the mobile RFID circumstance except that taking place in com-panies or retail shops which try to collect personal information. Main assumptions are as follows: - Privacy in the mobile RFID circumstance comes into force when a person holds a tag attached thing and both
information on his/her personal identity (refer-ence number, name, etc.) and the tag (of commodity) are connected to each other. - Privacy protection information are concerned with the tag attached object (its name, value, etc.) and the personal identity (of its owner or reference). - When it comes to the level of access authority, the owner can have access to any personal information on his/her object, an authorized person (e.g., pharmacist or doctor in medical care) only to access permitted information, and an un-au-thorized person to nowhere. 4 Conclusion In this paper, the WIPI-based adult certificate method is proposed. This Paper is for checking a user whether he/she is an adult or not, when a user would like to read some adult-related data using mobile devices playing a role of tag readers on mobile RFID environments. Instead of current adult-certification method using one’s own mobile devices, an adult certification in this paper uses any mobile devices and provides user’s anonymity. In this paper, the application areas of the proposed platform are discussed briefly in the fields of RFID based LBS (Location Based Service), RFID-based mobile payment, RFID-based mobile CRM (Customer Relationship Management), and mobile ASP (Applications Service Provider). For further study of this area, the verification and validation of the light-weight security middleware model by design and implementing some pilot-scale service system is necessary. It is also required to develop evaluation and authentication methodologies with the assistance of toolkits for the granularity of the QoS (Quality of Service) of the pilot-scale service system.

Trust Management for User-Centric Identity Management on the Internet ★

Daeseon Choi (ETRI, Korea); Seung-Hun Jin (ETRI, Korea); Hyunsoo Yoon (KAIST, Korea)

This paper proposes a trust management scheme for an application service provider to evaluate trustworthiness of a credential provider in the environment of user-centric identity management. In user-centric identity management scheme, a user selects the credentials that manage his identity and provide the identity information and authentication credential to the service providers. In contrast with the conventional identity management scheme in which the service provider selects credential provider and makes a business agreement about long term trust relationship, use-centric identity management scheme requires the service provider to evaluate the trustworthiness of the credential provider that issues the credential of a user. In this paper we propose a reputation based scheme that makes it possible for service provider to evaluate trustworthiness of credential issuer when a user submits some credential. In this scheme, the trust of credential provider is classified for kind of information that the credential can carry. Possible kinds of information can be authentication assertion, personal attribute information, and location information. To evaluate the trustworthiness of credential provider, someone has to collect and evaluate the information required for the providers and manage the evaluation result and provide the result to the service providers. It is unrealistic that there is a single authority that can evaluate all class of trust information. So we propose reputation based evaluation method. In our scheme, the evaluation is made based on the reputation of the service providers that have experience of credential providers. With our scheme, evaluation can be made effectively and safely. That makes possible free of selection by users. It is the core of user centric identity management.

12:30 PM - 1:30 PM

Keynote 4 ★

Effects of Energetic Particle Radiation on High Definition Video Imager Pixel Loss
Room: Trinity 5

A Microcosm of Change: The Impact of the Videogame Industry on Social Networking and the Consumer Media Experience ★

Corey Carbonara (Baylor University, USA)
by Dr. Corey P. Carbonara, Baylor University Today, digital media has provided other means of placing entertainment, news and information in our possession, creating, in fact, a new social networking phenomenon that has had a major impact on our society. When examining the role of consumer electronic devices in the home, there is a clear migration from the first generation of analog devices for human communication and entertainment--consisting of rotary phones, rabbit ears and transistor radios--to a more sophisticated set of digital devices that have more functionality and multiple uses. What has allowed digital devices to promulgate includes the following developments: (1) the development of a network of networks--affecting the way we work, learn and play; (2) plentiful and cheap data storage; (3) digital displays and imaging devices; and (4) computing/communication capabilities that impact nearly every digital device. Specifically, we know these devices by their function as digital audio/video storage, videogame consoles, personal digital assistants, cell phones, media hubs, digital video recorders, HD displays, network routers, digital cameras, etc. This paper will focus on the impact of current developments in the videogame industry as a case study representing massive changes in society due to the direct impact of new consumer digital technologies in the marketplace. The method used in this case study will be
an examination of the recent trends in the videogame industry from a contextual perspective—analyzing the current technological, economic and socio-cultural issues and impacts that have shaped its growth. Videogame technology today represents an amalgam of major advancement in wireless, data storage, and processing technologies. From the rise of massively online virtual worlds to the power of the next generation consoles capable of parallel processed multi-core computing with wireless interfaces allowing for seamless connectivity and an emphasis on HD experiences, videogames have the potential to change the way we establish community—actually placing consumer electronics at the nodal point of social change. A close examination of this industry gives us a powerful lens to track major changes in the use of digital technologies for working, learning and playing. The videogame industry now accounts for tens of billions of dollars annually and spans the PC, console and mobile environments. The average age of a gamer is now 30 years old, nearly half of these are women. In North America alone, hundreds of millions of videogames have been purchased, with half of all households on track to own two or more consoles by the end of the decade. New virtual worlds have attracted millions of subscribers, creating true economic value within these “metaverses” that have rivaled the GNP of actual countries. The use of a virtual online world as a magnet for social networking brings with it a potential for new frameworks of fluid and dynamic learning—using videogames as a catalyst for advanced media-based simulations and the development of a new industry of serious games. It is the purpose of this paper to track the latest developments within the videogame industry and discuss the impact it is having on the consumer electronics field as a whole.

1:30 PM - 3:00 PM

Audio Video Technology 7

Room: Trinity 6-8  Chair: Nasser Kehtarnavaz (University of Texas at Dallas, USA)

System Aspects of TV-Anytime Metadata Codec in a Uni-directional Broadcasting Environment

Minje Kim (Electronics and Telecommunications Research Institute, Korea); Minsik Park (ETRI, Slovakia); Yang Seung-Jun (ETRI, Korea); Ji Hoon Choi (ETRI, Korea); Han-kyu Lee (ETRI, Slovakia)

Digital broadcasting environment in which digitalized AV contents can be sent has a room for data transmission as well. In this circumstance, effective encoding method of verbose XML type TV-Anytime metadata becomes important and also the broadcasting environment-specific refinement of TV-Anytime transportation specification is in great need. This paper suggests a well-tuned framework for encoding TV-Anytime metadata. Our TV-Anytime codec, the main subject of this paper, includes revision of the whole TVA codec system specification and external programming interfaces to communicate with the external modules in our uni-directional broadcasting head-end system.

Ji Hoon Choi received the B.S and the M.S degrees in electronic engineering from Kyunghee University, Suwon, Korea, in 1999 and in 2001, respectively. In 2001, he joined the Data Broadcasting Research Group in ETRI(Electronics and Telecommunications Research Institute, Deajeon, Korea, where he have been working on the interactive data broadcasting system. His research interests are the data broadcasting and network QoS.

Optimization and Implementation for AVS Audio Decoder on RISC Core

Lei Bai ying (Zhejiang University, P.R. China)

This paper proposes the software optimization to implement AVS-P3 (audio,version1.0) realtime on ARM 920T (S3C2440). The optimization for embedded systems has been recognized as a key issue in improving cost as well as performance. In this work, we describe our experience in optimizing AVS audio decoder based on ARM9. We mainly optimize the inversed IntMDCT, CBC and dequantization modules. The test result shows that the decoding speed improves 10 times than the original AVS audio decoder decoding the 128kbps bitrate, 48 KHz sample rate, stereo wave signal, meanwhile, the memory space is reduced from 130.78KB to 59.91KB, achieved 54.2% improvement.

Software Optimization and Realtime Implementation of AVS-M on RISC Core

Lei Bai ying (Zhejiang University, P.R. China)

AVS-M is the recent mobile video coding standard of China. Currently, ARM cores are widely used in mobile applications because of their low power consumption. In this paper, a scheme of the AVS-M decoder realtime implementation on 32 bit MCU RISC processor ARM920T (S3C2440) is presented. The algorithm, redundancy, structure and memory optimization methods to implement AVS-M realtime are discussed in detail. The experiment results demonstrate the success of our optimization techniques and the realtime implementation. The ADS, MCPS and simulation results show that the proposed AVS-M decoder can decode the QVGA image sequence in realtime with high image quality and has low complexity and less memory requirement. AVS conformance test result
confirms the proposed AVS-M decoder full compliance with AVS. The proposed AVS-M decoder can be employed in many real-time applications in the third generation communication.

Consumer Networks 2

Room: Trinity 1-3 Chair: Jaber Khoja (The University of Texas at Arlington, USA)

**Design and Implementation of Home Intercom and Security Control System for Buildings**

Y. Bai (Fu Jen Catholic University, Taiwan); Shi-Chang Chen (Fu Jen Catholic University, Taiwan)

With the size of buildings getting larger, the efficiency and reliability of home intercommunication is getting more challenged, and the home intercom and security systems for buildings need to be redesigned. There are several kinds of these systems, such as, the traditional analog system, the industry network, the PSTN network, the Ethernet and the wireless LAN. The signals of the traditional analog system have to be transmitted through the analog circuit. Thus the loading and the complexity of the circuit depend on the number of users, making the traditional analog system suitable only for small-scale buildings. Since signals are transmitted in a digital format which is easy to do various kinds of processing, this paper proposes a design serving more than a thousand users in a building by using a kind of industry network communication system. Our design provides a low-cost, simple architecture with a high stability solution for home intercom. A home intercom and security communication system can be divided into talk-back, image monitors and security monitors, so we only need to use application layer, data link layer and physical layer of the OSI/ISO standards. We adopt the RS-485 communication network whose wire connection has the advantage of low cost and is easy to build. The main wire and the room wire can be installed with additional isolation to protect the main wire from damage. The proposed design of the communication protocol needs to consider the number of system nodes. For a building with over one thousand users, cyclic polling or multiple polling does not provide a good performance. As cyclic polling wastes time and resources, we adopt a structure that mixes “Peer to Peer” and “master to slave” to fulfill the requirements of the specification. For example, if there is an alarm and another user needs to be notified, the Peer to Peer communication protocol is faster. However, the home intercom system’s main purpose is to transmit a short package which provides control and status signals. Therefore, we use UART (Universal Asynchronous Receiver and Transmitter) whose simple data format is 1 start bit + 8 data bits + 1 stop bit. The transmission interval among each byte is 2 bits. The structure of the package consists of start byte, body and stop byte. The data link layer can use back-off, control frame or policy-base network to obtain the right to use a channel. Our intercom and security system can be used in large buildings as it provides a better solution for a large number of address members. Our mapping relation uses a two-dimensional array between the logic address and the user address. The user address generally includes user number, floor number and room number.

Ying-Wen Bai is a professor in the Department of Electronic Engineering at Fu-Jen Catholic University, Taiwan. His research focuses on mobile computing and microcomputer system design. Ying-Wen Bai obtained his M.S. and Ph.D. degrees in electrical engineering from Columbia University, New York, in 1991 and 1993, respectively. Between 1993 and 1995, he worked at the Institute for Information Industry, Taiwan.

**A RFID-Enabled Privacy and Security Preserving Ubiquitous Healthcare System**

Jieun Song (Electronics and Telecommunications Research, Korea, Korea)

Recently, the stream of electronics healthcare system has moved to ubiquitous healthcare system, which supports more correct, timeless, valuable healthcare services with intelligent system such as RFID technology or smart card or medical sensors. The ubiquitous healthcare service means that users can utilize healthcare services naturally and conveniently as a part of their daily routine, regardless of place or time. In ubiquitous healthcare service environment, users including patient are able to use various healthcare services anytime, anywhere and through any devices. It makes users’ lives more comfortable, efficient and healthy. To expand the healthcare service area, there have been many researches and developments for new medical technologies and systems such as wearable or implantable bio-medical sensor, RFID (Radio Frequency Identification) technology, high performed healthcare information system and so on. Although the research and development for the application of healthcare systems are processed actively, however the healthcare systems still have some security weaknesses such as medical data security and patient privacy provision problems. As it is able to collect the patient medical data from a variety of devices, wired or wireless medical sensor, it happens to trace a patient and collect his private medical data illegally. In addition, the more different healthcare institutes are networked and co-work each other, the more medical information is shared and transferred through internet. Online collection and processing of personal data also result in serious threats to privacy. Therefore, much attention and origin considerations based on characteristics of u-healthcare network must be paid to the privacy principle for safety provision including the confidentiality, integrity and availability of information. We analysis and discuss the healthcare-specific issues and countermeasures for privacy and security in RFID-enabled healthcare system. RFID technology has been utilized actively for more evolved ubiquitous computing service. And we will design and propose privacy and security
Usability Enhanced Privacy Protection Scheme based on Users' Responses

Han-Gyu Ko (UST, Korea); Seung-Hyun Kim (Electronics and Telecommunications Research Institute, Korea); Seung-Hun Jin (ETRI, Korea)

Recently, the concern and significance about the privacy protection for Internet users increase as many incidents of privacy violation have been occurred including social security number theft, personal information leakage from public home pages and so on. Regarding the privacy problem, there are a few traditional privacy protection schemes. One of the representative schemes is setting privacy policies of user's private information. Well-known privacy protection methods, such as P3P, APPEL, XACML, belong to this category. However, there are some shortcomings of the privacy protection schemes that make a user to set his/her policies. First of all, privacy policy setting is too static to reflect timely-variant user's privacy preferences. Because the user's privacy preferences are changed according to various situations, there will be many exceptions that the static definition of the privacy...
policies cannot cover. Second, privacy policy setting is a kind of cumbersome work to most of users. Users who want to set their own policies should know how to use the corresponding privacy policy definition language. Moreover, in order to reflect users’ preferences on the privacy policies, they should consider sensitivity of each personal information and purpose of each personal information usage. These shortcomings decrease the usability of the policy based privacy protection methods. In this paper, we propose a usability enhanced privacy protection system based on user's response. It provides two of usability enhanced functionalities as following. First, our system reflects user’s privacy preferences by profiling their responses. Whenever there is a request for personal information of a user, our system sends the user a query message which asks the user's agreement about use of requested personal information for given purpose. Our system uses the user's response to measure and reflect his/her privacy preference as well as to decide whether releases the request information. There is one privacy preference table for each user. Each preference is represented a vector value that is a result of profiling the user's response. According to the vector values, our system reflects each user's privacy preference. It means that the users can control their personal information and reflect their privacy preferences without any privacy policy definitions. All they need is just to input their responses as 'Yes' or 'No'. In addition, our system considers data dependencies of the requested information to notice the expected consequences if the user allow release of the personal information. We define a table of data dependencies and use it to create the warning message that will be included in a real-time query message. Therefore, Internet users can make appropriate decisions even they cannot predict the consequences by themselves.

Cajetan Akujuobi (Prairie View A&M University, USA); Nana Ampah (Prairie View A&M University, USA); Matthew Sadiku (Prairie View A&M University, USA)
The need to use quantitative methods to detect intrusion is increasing due to the high false positive and false negative rates of existing Intrusion Detection Systems (IDS) and Intrusion Prevention Systems (IPS). Most network security techniques employed by the IDS and IPS depend mainly on packet behavior for detection. This work applies a quantitative approach based on Maximum A Priori (MAP) detection rules with the hope of reducing the high false positive and false negative rates. The entire system has been represented by a mathematical model of a discrete binary communication channel having two possible input messages and two possible output symbols. The network under study is assumed to have only one entry point (sender) for now, with a number of nodes (receivers). Also, all normal operational packets are referred to as normal packets and any other packets are referred to as abnormal packets. The analysis strategy used here is anomaly detection. The developed algorithm initially calculates the a priori probabilities for the normal and abnormal packets both at the sender and entry ends. These values are used as threshold probabilities to be compared with probabilities of future incoming packets. MATLAB was used in coding the developed algorithm. This work will be expanded by modeling the entire system as a continuous binary communication channel and also by considering multiple entry points as future works, with the intention of improving the results obtained so far.
Dr. Akujuobi is the founding Director of the Center of Excellence for Communication Systems Technology Research (CECSTR) at Prairie View A&M University (PVAMU). He is also the founding Director of the Analog Mixed Signal (AMS), DSP Solutions and High Speed (Broadband) communication Programs. He is a Professor of Electrical Engineering and Head, Engineering Technology Department. Dr. Akujuobi is the Texas Instruments University Relations Manager at Prairie View A&M University. He belongs to many professional organizations such as IEEE (Senior Member), ISA (Senior Member), ASEE, SPIE, and Sigma Xi, the Scientific Research Society. Dr. Akujuobi has over 25 years experience in engineering education, research and development. His current research interests include wavelet-based research, mixed signal systems, DSP Solutions, High-Speed (Broadband) communication systems, signal and image processing using such tools as wavelet and fractal transforms. Dr. Akujuobi has published extensively and has taught as a university faculty and researcher in numerous private and state universities. He was a participant and collaborative member of ANSI T1E1.4 Working Group which has the technical responsibility during the development of T1.413, Issue 2 ADSL Standard.

Application of Wavelets and Self-similarity to Enterprise Network Intrusion Detection and Prevention Systems ★
Cajetan Akujuobi (Prairie View A&M University, USA); Nana Ampah (Prairie View A&M University, USA); Matthew Sadiku (Prairie View A&M University, USA)
Securing Enterprise networks has so far been considered under two broad topics (i. e. Intrusion Detection Systems - IDS and Intrusion Prevention Systems - IPS). So far, there is no algorithm, which guarantees absolute protection for a given network from intruders. Most existing IDS and IPS techniques introduce high false positive and false negative rates, which need to be eliminated or reduced considerably. This paper will concentrate on network packets behavior leading to network-based intrusion detection. It will employ anomaly detection as its analysis strategy. In the field of signal analysis, the methods of wavelet transform have gotten wide application because of its unique merit. The wavelet idea will be used in this paper to better enhance and improve the novel security
algorithm developed in this research project. The self-similarity property of real network traffic will be used together with the signal detection abilities of wavelets in detecting attacks. The technique used here will also try to reduce the effectiveness of distributed attacks, which deny authorized users access to system resources. Securing of all network security data, which is an important limitation to existing IDS and IPS is ensured by this technique. The results have been very successful.

Dr. Akujuobi is the founding Director of the Center of Excellence for Communication Systems Technology Research (CECSTR) at Prairie View A&M University (PVAMU). He is also the founding Director of the Analog Mixed Signal (AMS), DSP Solutions and High Speed (Broadband) communication Programs. He is a Professor of Electrical Engineering and Head, Engineering Technology Department. Dr. Akujuobi is the Texas Instruments University Relations Manager at Prairie View A&M University. He belongs to many professional organizations such as IEEE (Senior Member), ISA (Senior Member), ASEE, SPIE, and Sigma Xi, the Scientific Research Society. Dr. Akujuobi has over 25 years experience in engineering education, research and development. His current research interests include wavelet-based research, mixed signal systems, DSP Solutions, High-Speed (Broadband) communication systems, signal and image processing using such tools as wavelet and fractal transforms. Dr. Akujuobi has published extensively and has taught as a university faculty and researcher in numerous private and state universities. He was a participant and collaborative member of ANSI T1E1.4 Working Group which has the technical responsibility during the development of T1.413, Issue 2 ADSL Standard.

Workshop 2 ★

Using Interaction Design and Listening Skills to Create Products That Your Customer Will Love

Room: Trinity 5

Security & Digital Rights Management 4 ★

Room: Trinity 4

**Practical Data-Hiding in MPEG-4 AAC Audio ★**
Xu Shuzheng (Tsinghua University, P.R. China)

This paper describes a novel data-hiding scheme based on scale factor modification in MPEG-4 AAC file. The capacity of hiding data in the file is analyzed and the impact on AAC quality is evaluated. The basic idea behind the scheme is as followed: by altering the quantization and coding process through modifying the scale factor so as to produce more bits, then the redundant bits will be substituted by hiding data. The qualities of data-hiding AAC signals were strictly assessed MUSHRA listening test. The test results show there is no statistically significant degradation between the original MPEG-4 AAC encoded items and the covered ones, no matter that the hiding data capacity varies from 10% to 100%. The spectrum and size incensement of original and covered AAC signal are also analyzed with detail results which will be shown in the full manuscript. The scheme is flexible in the capacity of hiding data and can obtain good balance between security, robustness and quality. Experimental results confirm the potential of this new scheme and prove it to be an attractive solution in intelligent property right protection and covert communications.

**Service and Content Protection in Mobile Multimedia Broadcast ★**
Teodor Buburuzan (Technical University of Braunschweig, Germany); Gunther May (Technical University of Braunschweig, Germany); Khaled Daoud (Braunschweig Technical University, Germany, Germany)

Abstract— In order to enable a new set of services that go beyond mobile TV, both the DVB Project and the OMA have developed two, new frameworks designed for Service and Content Protection. Both frameworks were basically designed to provide the same thing, and this is a standardised service provisioning platform. But because these specifications stem from different parts of the telecommunication world, there are also some differences to be noted.

I was born in Iasi, Romania on July 27, 1981. In 2000, after my high school diploma, I started studying Computer Science at “Al. I. Cuza” University of Iasi. Between April 2003 and May 2004 I was in Germany at Technical University of Braunschweig within the SOKRATES / ERASMUS scholarship program. During this time I worked as student assistant at the Institute for Computer applications in Civil Engineering and the Institute of Computer and Communication Network Engineering. Back in Romania, in June 2004 I got my Computer Science degree with the thesis “Mobile Agents in Network Discovery Services” and an overall average grade of 10 (max 10). Between September 2004 and January 2005 I worked as a student scientist at the Institute for Communications Technology, Technical University of Braunschweig and since February 2005 I am acting there as a full scientist. My main research interest lies in the area of hybrid and heterogeneous communication networks, work in the frame of DAIDALOS and DAIDALOS-II EU integrated projects.
A H.264 based four-way secured Video Conferencing System with Encryption and Watermarking

Tanushyam Chattopadhyay (Tata Consultancy Services, Kolkata, India); Chirabarta Bhaumik (TCS, India); Arpan Pal (TCS, India)

Last few years have witnessed rapid growth in video coding technology. Among various standards, H.264/Advanced Video Codec (AVC) is found to be of significant importance regarding reduced bandwidth, better image quality and network friendliness. One of the current fields of interest is to develop a system with authentication and copyright protection methodology embedded within an efficient video for an H.264 based Video Conferencing System. With the advent of digital computers, high-speed computer networks, Internet and wide spread use of World Wide Web have revolutionized the way digital data are distributed. The wide spread accesses to multimedia objects and the ease of copying digital data without loss of considerable fidelity have motivated the study of digital watermarking techniques in the last few years. As a result of that a large number of publications on image, video, audio watermarking techniques can be found in the literature in the last decade [1, and references there in]. Digital watermarking is a well-known method, which is used to ensure digital right managements in multimedia data. The main purpose of digital watermarking is to embed digital copyright information imperceptibly and robustly in the cover media. In general the embedded data contains information about the origin, ownership, destination, copy control, transaction information etc. Common application of digital watermarking includes transaction tracking, copy control, authentication, legacy system enhancement, database linking etc. Growing popularity of video based applications such as Internet multimedia, wireless video, personal video recorders, video-on-demand, set-top box, videophone and videoconferencing have a demand for much higher compression to meet bandwidth criteria and best video quality as possible. Different video Encoder Decoders (CODECs) have evolved to meet the current requirements of video application based products. Among various available standards H.264 / Advanced Video Codec (AVC) is becoming an important alternative regarding reduced band width, better image quality in terms of peak-signal-to-noise-ratio (PSNR) and network friendliness [2], but it requires higher computational complexity. In this paper we present a secure video conferencing system using H.264. The system not only provides security in terms of encryption, but it also provides copy control enabled conferencing session storage (that can be used as Minutes of the Meeting) through watermarking. The hallmark of both the encryption and watermarking technique is that both of these are performed in the H.264 compressed domain using less computational power.

A Survey on video watermarking attacks with focus on watermarked compressed H.264 stream

Tanushyam Chattopadhyay (Tata Consultancy Services, Kolkata, India); Anshuman Goswami (TCS, Kolkata, India); Arpan Pal (TCS, India)

Last few years have witnessed rapid growth in video coding technology. Among various standards, H.264/Advanced Video Codec (AVC) is found to be of significant importance regarding reduced bandwidth, better image quality and network friendliness. One of the current fields of interest is to develop a system with authentication and copyright protection methodology embedded within an efficient video codec. At the same time some evaluation for the watermark should be there so that the watermarking technique can be tested. In this paper we first performed a survey on available attacks on watermarking techniques, then discussed about the attack specific to H.264 video watermarked data and finally discuss a measure of goodness for the evaluation of the watermarking scheme. High speed computer networks, the Internet, efficient DVD/VCD recorders & low-cost storage media have revolutionized the way in which multimedia content is distributed. Applications like multimedia streaming, video-on-demand, videoconferencing, IPTV, personal video recorders, set-top boxes and the like have become very popular mainly due to the combined effect of advanced codecs like H.264/AVC [9] coming to the market and improved bandwidth ushered in by new network technologies. However this has been accompanied by a growing threat of IP theft & illegal redistribution, thus entailing massive revenue losses. One of the means to combat this threat could be through digital watermarking. Digital Watermarking is a technique of embedding additional data into the original or host data directly and imperceptibly such that the watermarked data can be extracted later to make an assertion about the original content. Typical applications of digital watermarking include copyright protection, distribution tracing, authentication and conditional access control [2]. However, as the host data has to go through certain amount of signal processing before reaching the viewer either as a part of the standard signal flow or as part of a attack designed to destroy/modify the watermark, the watermark must be resilient to these transformations which can be thought of as attacks. A benchmark or evaluation scheme must be in place to test the goodness of a particular watermarking technique. This will give a measure of how robust the watermarking scheme is to several forms of intentional and unintentional attacks. Some benchmarking schemes are already available to evaluate image watermarking, like the one described in [7]. However, to develop a good evaluation scheme for video watermarking, a survey on available video watermarking techniques is very useful, such as in [42]. The video watermarking technique essentially depends on the encoding scheme used. Of the available video watermarking techniques, very few deal with H.264/AVC compressed video stream. The reason for laying such emphasis on H.264/AVC over the numerous other video codecs has to do primarily with the fact that there is sufficient reason to consider this codec as “the codec” for the future, [8]. H.264/AVC offers enormous
reduction in terms of bandwidth with much better peak-signal-to-noise-ratio (PSNR), as well as, much improved network friendliness. Though it requires higher computational complexity, novel optimization techniques like the one proposed in [43] can take care of this issue. In this paper, we study the existing video watermark attacks with a focus on H.264/AVC compressed video stream. We will see that not all of these attacks are applicable to H.264/AVC. We will also see how important it is to develop a good evaluation scheme not only to test the robustness issues of a watermarking technique, but also to give inputs as to how the watermarking scheme can be improved. We will go on to illustrate an attack technique specific to a H.264/AVC compressed video watermarking scheme developed by us. We believe that this will give an insight into what could be done to improve our watermarking technique and maybe others’ specifically targeted towards H.264/AVC application.

There is a strong notion going around amongst the multimedia community that future applications will show a trend to migrate to H.264/AVC from MPEG series and in this context, we believe that our work will prove useful in managing Digital Rights Management (DRM) issues for future video applications. Tanushyam Chattopadhyay, (member IEEE and ACM) has received the BSc in Physics from Visva Bharati Santiniketan and completed his Master in Computer Application(MCA) from Bengal Engineering College, Shibpur, India, in 1998 and 2002, respectively. He was awarded with the University Gold medal in MCA. He started his career as research personnel in Indian Statistical Institute, Kolkata, and later on, joined the software professional in Research and development section of Embedded Systems group of Tata Consultancy Services Limited and still working as an IT analyst. He is one of the key programmers involved in the development of a secured H.264 based video conferencing and video telephony system. His areas of interest include video compression, digital watermarking and encryption, video segmentation, Video Search engine development and video summarization.

**A Novel Scheme for evaluation of Video Watermarking Technique against Attacks**

**Tanushyam Chattopadhyay** (Tata Consultancy Services, Kolkata, India); **Arpan Pal** (TCS, India)

In the current trend of Internet based multimedia applications, digital rights management and security is an essential requirement for authentication and copyright management. There are different watermarking techniques available in the market and at the same time lots of attacks have been developed to destroy any hidden information in terms of watermarking embedded into the video stream. So it is extremely important to have a system that can report the goodness of any watermarking scheme in terms of its robustness against attacks. But there is hardly any work that can be found in the area of video watermarking. In this paper we present a novel technique where we develop a measure that can evaluate an attack by comparing the attacked video with the original watermarked video stream. Usually any binary image or any text message is used as embedding information during watermarking. In our approach we compare inserted binary image and/ or text message with retrieved image and/ or text message using multifactorial based approach. We compare the score with the results obtained by Mean Opinion Score (MOS), which is purely based on Human Vision Psychology (HVS), and finally assigning a fuzzy membership value to each class of watermark. In the age of multimedia and Internet a number of Internet based multimedia applications like IPTV, video conferencing system, videophone has been developed. Lots of work has been done in the field of digital communication to provide greater bandwidth for communication. At the same time last decade has witnessed a huge growth in the research of multimedia, more specifically in the field of video codec to encode video stream with lesser number of bits yet generating better image quality. Now digital storage and Internet based multimedia applications in video industry are threatened due to illegal and unauthorized copy of multimedia data. This increases the need for devising copyright protection and authentication measures. As a consequence of this requirements different water marking techniques come into focus. Any watermarking scheme can be evaluated by its performance measured in terms of its complexity and robustness against attacks. Any attack to watermarking system is the technique to remove or change the hidden data in the video bitstream. So in the study of watermarking, it is essential to concentrate on attacks on watermarking scheme, too. Because it is used to simulate the process of any end user trying to remove or destroy the hidden information embedded into the video stream as the security measure taken during watermark embedding. Moreover it is also a measure of goodness of watermarking scheme in terms of robustness of the scheme. In the current paper we device a measure that can be used to evaluate the attack as well as the watermarking scheme. Here we evaluate the watermarking scheme and the attacking methods in three ways. First we compare the attacked stream with watermarked stream. A total of seventeen odd parameters are used to measure the goodness of video quality, which are then unified using multifactorial approach. Twenty users (15 men and 5 women), who are asked to rate the attacked video quality as per their perception into 4 classes, judge the attacked streams. Samely the retrieved image and/or text message using multifactorial based approach. We compare the score with the results obtained by Mean Opinion Score (MOS), which is purely based on Human Vision Psychology (HVS), and finally assigning a fuzzy membership value to each class of watermark. In the age of multimedia and Internet a number of Internet based multimedia applications like IPTV, video conferencing system, videophone has been developed. Lots of work has been done in the field of digital communication to provide greater bandwidth for communication. At the same time last decade has witnessed a huge growth in the research of multimedia, more specifically in the field of video codec to encode video stream with lesser number of bits yet generating better image quality. Now digital storage and Internet based multimedia applications in video industry are threatened due to illegal and unauthorized copy of multimedia data. This increases the need for devising copyright protection and authentication measures. As a consequence of this requirements different water marking techniques come into focus. Any watermarking scheme can be evaluated by its performance measured in terms of its complexity and robustness against attacks. Any attack to watermarking system is the technique to remove or change the hidden data in the video bitstream. So in the study of watermarking, it is essential to concentrate on attacks on watermarking scheme, too. Because it is used to simulate the process of any end user trying to remove or destroy the hidden information embedded into the video stream as the security measure taken during watermark embedding. Moreover it is also a measure of goodness of watermarking scheme in terms of robustness of the scheme. In the current paper we device a measure that can be used to evaluate the attack as well as the watermarking scheme. Here we evaluate the watermarking scheme and the attacking methods in three ways. First we compare the attacked stream with watermarked stream. A total of seventeen odd parameters are used to measure the goodness of video quality, which are then unified using multifactorial approach. Twenty users (15 men and 5 women), who are asked to rate the attacked video quality as per their perception into 4 classes, judge the attacked streams. Samely the retrieved and embedded binary images and texts are compared using some statistical parameters. Finally we conclude about the method in terms of these three parameters. In section II we review the basics of video watermarking techniques. Section III discusses different parameters used in evaluating watermarking scheme. In section IV we discuss the technique of evaluation of the attack scheme. Some results obtained by attack on a H.264 based watermarking scheme developed by TCS is described in section V. Finally the conclusions are drawn in section VI by discussing the results obtained by applying attacking schemes to the watermarking scheme developed by TCS. Tanushyam Chattopadhyay, (member IEEE and ACM) has received the BSc in Physics from Visva Bharati.
CryptMark: A Novel Secure Invisible Watermarking Technique for Double Layer Protection of Color Images

Saraju Mohanty (University of North Texas, USA)

Title: CryptMark: A Novel Secure Invisible Watermarking Technique for Double Layer Protection of Color Images

Abstract: With the explosive growth of internet technology, many innovative applications requiring exchange of large amounts of multimedia data have become feasible. However, this kind of convenience with which authorized users can access information, turns out to be a mixed blessing because of information piracy. The emerging field of digital rights management (DRM) systems addresses the issues related to the intellectual property rights of digital content. In this paper, we present a novel invisible watermarking method that simultaneously uses cryptographic and watermarking methods to provide a double-layer protection to the digital media; this approach can be an effective technique for DRM. Our proposed method securely hides binary information in color image media, and securely extracts and authenticates it using a secret key. Experimental results prove that our proposed invisible watermarking techniques are resilient to 90% of the well known benchmark attacks and hence a fail-safe method for providing constant protection to ownership rights. If the color host image is the image of a passport holder and the binary information is biometric, such as fingerprints or an iris image, our proposed algorithm can be used to hide the biometric information in the passport holder’s image securely. Hence it can be used in electronic passports (e-passports). Thus, the proposed algorithm can have immense impact is solving homeland security and border security issues through the use of e-passports. Various aspects of content management such as content identification, storage, representation, and distribution and intellectual property rights management are highlighted in DRM. Unauthorized access of digital content is being prevented by implementing encryption technologies. However, it does not prevent an authorized user from illegally replicating the decrypted content. Hence, encryption alone does not address all the IP issues related to DRM. Digital watermarking can be used for establishing ownership rights, tracking usage, ensuring authorized access, preventing illegal replication and facilitating content authentication. Therefore, a two layer protection mechanism utilizing both watermarking and encryption is needed for effective DRM; the contribution of this paper is to provide such a framework. The paper presents a novel invisible watermarking method that uses cryptographic and watermarking methods simultaneously to provide a double layer protection to the digital media which can be an effective technique for DRM. Our proposed method securely hides binary information in color image media, and securely extracts and authenticates it using a secret key. The advantage of encrypted watermark processing is that at no point in time raw watermark information is passed in the transmission channel, thus providing maximum security. The proposed embedding process uses both DC and AC DCT (discrete cosine transform) components to carry the payload, unlike most of the existing algorithms which rely heavily on low frequency AC components. This provides more resilience to lossy compression, a process that is heavily dependent on smaller low frequency values of AC components. In addition, we selectively add or subtract the watermark from the DCT coefficients instead of only performing an adding operation as in most available algorithms. Our approach allows carrying maximum payload with highest robustness and highest undetectability, the three contradictory objectives of data hiding mechanisms.

3:15 PM - 4:00 PM

A/V RF & Wireless 3⭐

Room: Trinity 6-8

An E-Narrator System Design and Implementation ⭐

Chin Lin (Cheng Kung Univ., Taiwan)

MP3 is recognized as popular handy consumer electronics for personal audio recording for music or voice. MP3 can be used to edit and record narrating message under certain format and protocol. An MP3 is combined with radio frequency (RF) in microprocessor control to activate and select specific section of audio narration for general applications. The exhibition is installed with RF transmitter in directional interrogation to send the assigned code in digital format. The E-Narrator device contains a microprocessor to control and manage MP3 playing. The receiving RF code is converted into address of the audio record in the MP3 activating by RF receiver. The RF receiver on the
E-Narrator receives pronouncing code from the wall RF transmitter. Sequential or non-sequential MP3 selections are designed and controlled into microprocessor software to identify the prompt search for next narration. The Compact Flash memory on the MP3 can be updated at the entrance of the exhibition hall, if the E-Narrator device is owned personally, or can be refreshed by the Exhibition Administration for E-Narrator rental. In the E-Narrator system design, the RF transceivers are tuned to catch the RF signal in directional short range to avoid interference. The proposed E-Narrator system has the following characteristics. (1) It is a very compact design, handy operation, and low cost to own one device. (2) The RF transmitter is easy installation by dip switch and low cost for investment. (3) The proposed E-Narrator device can owned as personal MP3, only to install the exhibition narration. This E-Narrator can also be a rental device at the exhibition box office for any type of exhibitions that require narration. (4) The narration record is easy to prepare and edit onto the flash memory. In the tests, the E-Narrator device can last for more than 40 hours on Li-ion battery with flexible selection functions. It is quite useful to public exhibition applications for personal own device or rental device.

Prof. Lin was born in Chang Hua, Taiwan. He received BSEE and MSEE from Department of Electrical Engineering, National Cheng Kung University, Tainan, Taiwan, in 1975 and 1977, respectively. He received Doctor of Engineering from Department of Electrical Engineering, Lamar University, Beaumont, Texas in 1983. Dr. Lin is current professor in Department of Aeronautics and Astronautics, National Cheng Kung University. His research interests are control applications, avionics system, wireless data surveillance system, magnetic suspension system, and e-commerce system.

### Wireless and mobile communications determine our future!
Dr. Mohammed Anwar Rasheed (Canadian University, Sweden)
In these papers, the development of conduct plan that utilizes wireless communication depending on the spectrum management. Appling hot points wireless service propagation stations related with various aspects of spectrums. Almost every agency of the Government uses the spectrum in performing mandated missions therefore specifying spectrums for each establishment, ministry, company, university, school, house, factory, office...etc. is the intended objective. Also electromagnetic waves propagate outward in all directions so that extend it’s bandwidth by scatter the wavelength then split each wave in to the tiny numbers of waves to employ it in different functions. People wishing to use radio communication devices in a given area must cooperate if they are to avoid interference problems. If they operate on the same frequencies, at the same time and in the same area, their transmissions will produce interference in each other's receivers. Each user, in effect, prevents other simultaneous, nearby uses of a portion of the spectrum while transmitting. National Telecommunications and Information Administration (NTIA) has identified spectrum management objectives to guide Federal users of the radio spectrum. These objectives are similar in intent to the Act's guidelines and state that the Federal agencies are to "make effective, efficient, and prudent use of the radio spectrum in the best interest of the Nation, with care to conserve it for uses where other means of communication are not available or feasible." NTIA interprets the standard "effective, efficient, and prudent," and the reference to "the best interests of the Nation" as encompassing the overall benefits the American public derives from radio communication services, both Federal and non-Federal, as well as the needs of various Federal users and choices among competing users.

### Performance Analysis of an Adaptive Association Admission Control in the BcN Wireless LAN Access
Hyun-Woo Lee (ETRI(Electronics and Telecommunications Research Institute), Korea); Won Ryu (ETRI, Korea); Byung Sun Lee (ETRI, Korea)
This paper generally relates to the security of wireless local area networks(WLANs), and more particularly, to AP selection and access control methods for the performance of a station. In these days, IEEE 802.11 WLAN is widely deployed and used as an emerging service to connect high-speed Internet in the BcN (Broadband Convergence Network) wireless environment. But, if there are many users in hot spot area, they suffer a severe decrease of performance. Thus it needs an association and access control mechanism especially when it is used in the public environment. In this paper, we suggest an efficient association method using Beacon or Probe Response frames based IEEE 802.11 and analyze the performance of stations(STAs). Station selects AP using the information of the capability region in a Beacon or a Probe Response frame. According to the present paper, an association method for a WLAN access in the BcN, which includes a WLAN user terminal and an AP for relaying WLAN communications to and from the user terminal, includes the steps of the user terminal asking the AP's states to access with a radio channel; and the station selects and associates with the AP. Further, it is possible to improve the efficiency of network management.

### Consumer Networks 3
Room: Trinity 1-3
Chair: Jaber Khoja (The University of Texas at Arlington, USA)
Security & Digital Rights Management 5

Room: Trinity 4    Chair: Saraju Mohanty (University of North Texas, USA)

Persisted content protection among the heterogeneous devices

Jihyun Park (ETRI, Korea); Yeonjeong Jeong (Electronics and Telecommunications Research Institute, Korea); Ki-Song Yoon (ETRI, Korea); Jaecheol Ryou (Chung-Nam National University, Korea)

Recently, digital content have been protected by several different DRM solutions, which increase the users’ inconvenience if they have the several heterogeneous devices. For example, there may be much kind of devices in digital home environment. Those devices may have the different DRM in one another. So, content protected by one DRM cannot be consumed with other devices of other DRM because of the lack of DRM interoperability. Possible solutions to overcome this problem are such that all needed DRMs are installed into a device, or DRM adaption is performed appropriately. The DRM adaption is considered more feasible solution since some devices have the limited hardware resource, and conflicts may be occurred among the DRM agents if they are installed in same devices. In this paper, we propose the persistent content protection scheme and show implemented prototype system, which can protect the content while content moves to other heterogeneous devices. Our prototype system consists of 3 consumer devices - PC, set-top box, and PMP. And these devices have the different DRM in one another. Moreover, these devices have the different characteristics. PC has the network capability and high computing performance. Set-top box also has the network capability, but computing performance is relatively low. PMP has not the network capability; instead, it can be connected to other devices via USB connection. And its performance is very low. We adopt the different content delivery method among these devices considering the device capabilities. Firstly, we describe our DRM adaptation scheme. It includes the content transcoding, device resolution, DRM adaptation, and so on. Next, we show the prototype system, and two different content delivery models. One is for delivery between PC and set-top box, and the other is for delivery between set-top box and PMP. We also describe some considering points for implementation. Finally, we show the some experiment result about DRM adaption overhead.

DRM content adaptation between different DRM Systems for Seamless Content Service

Do-Won Nam (ETRI, Korea); Yeonjeong Jeong (Electronics and Telecommunications Research Institute, Korea); Jihyun Park (ETRI, Korea); Ki-Song Yoon (ETRI, Korea)

DRM is a technology to protect and securely deliver content. Previous studies on this topic focused on security and encryption as a means of solving the issue of unauthorized copying, that is, locking the content and limiting its distribution to only purchasers. Current DRM provides protected contents to purchasers. It adopts a license-based mechanism which separates the keys from encrypted contents. The encrypted content is delivered to a player from a distribution server while the license including the keys is transported to the DRM client from a license server. After the contents are downloaded, users can freely use the contents if they have a license. However, although the users who get a license for the content, they can only use it on a specific portable audio or video device. For example, MP3 content protected with MS DRM can not be played on Apple’s iPod player because it doesn’t include the MS DRM client or module. So, it will be impossible to play one DRM’s content at another DRM’s portable audio and video device even within one user’s domain. The rationale of this paper comes from that all portable audio and video devices would not conform to one DRM. Currently many portable audio and video devices would remain conforming to many kinds of DRM standards or systems. Even after a DRM standard is broadly deployed, all portable audio and video devices would not conform to it. DRM adaptation is considered more feasible solution since some devices have the limited hardware resource, and conflicts may be occurred among the DRM agents if they are installed in same devices. For seamless content service, we propose the DRM content adaptation scheme that can be used to exchange content between different DRM systems. The proposed adaptation is for making one DRM system use the content which is protected from another DRM system. Usually there are three kinds of DRM data that covers a clear resource, content metadata and rights expression and they are packaged into a specific DRM content format. Since the main interest of this paper is how to adapt the DRM content between one DRM and another DRM, this paper establishes adaptation scheme which covers adaptation for resource, rights expression and metadata, content delivery, and key exchange. DRM content adaptation means that a specific DRM content is adapted into neutral DRM content. On one DRM side, he can get a clear resource, content metadata and rights expression from his DRM content and generate his own DRM content. And on another DRM side, he can get the clear resource, content metadata and rights expression from the neutral DRM content and generate his own DRM content. In this paper we propose a scheme that can adapts the DRM content of one DRM to that of another DRM in portable and video device environments. This paper is addressing solutions to the seamless content service for DRM content of portable audio and video devices, in particular, DRM content adaptation. DRM adaptation is for making another DRM system govern one DRM's DRM content, which is copied from one DRM system. Proposed adaptation scheme can serve as a bridging component to achieve secure transfer of DRM Content between different DRM systems.

http://edas.info/showProgram.php?_of__showProgram=&c=5136&form...bio%5D=1&program_view=everybody&action=Show+conference+program
Interlock System for DRM Interoperability of Streaming Contents

Do-Won Nam (ETRI, Korea); Jung Soo Lee (ETRI, Korea); Jeonghyun Kim (Electronics and Telecommunications Research Institute, Korea); Ki-Song Yoon (ETRI, Korea).

DRM (Digital Rights Management) is a tool for protecting copyrights and managing the rights of usage. Many DRM technology have been developed, which cause inconvenience for user in the usage of the content. DRM's standardizations have been proceeding for interoperability between different DRMs therefore. EXIM developed by Korea recently proposed the method for the interoperability of download contents, however, in the case of streaming content, it is difficult to apply EXIM technology for downloadable content because the process of encoding and transmitting of streaming content is complicated and the DRM technology is applied within these processes. In this paper, we propose the DRM interoperability structure for streaming content between the content providing server and the end device in which different DRM systems are installed respectively. We set up the interlock system for the DRM interoperability of streaming content between the content provision server and end device and using this system we wish to design the free and safe distribution system of the streaming content independent of the DRM types.

Jung-Soo Lee received his Ph.D. degree in Electronic Engineering from Hanyang University, Seoul Korea in 2005. Currently, he is a senior member of Electronics and Telecommunications Research Institute(ETRI). His research interests are Digital Watermarking, Fingerprinting, Image Processing and Digital Rights Management.

4:00 PM - 5:00 PM

Mobile Computing & Communication 1

Room: Trinity 6-8

SUIT: Scalable, Ultra-fast and Interoperable Interactive Television

Antonio Navarro (University of Aveiro, Portugal)

Abstract: The deployment of different types of networks makes available the contents anywhere and anytime. However, in the context of multimedia communications over heterogeneous networks, some interoperability related problems are unsolved so far. This paper discusses the advantages and disadvantages of SUIT (European IST project n. 028042) [1], a convergent architecture between two broadband wireless networks, WiMAX (16e) [2] and DVB-T/H [3]. The main idea is to combine in a fruitful manner DVB-T/H with the broadband wireless access network IEEE802.16e providing a low round-trip delay and robust communications at high mobile speed, above 150 km/h. In this context of network convergence, broadband networks like DVB-T/H and WiMAX will certainly play an important role, delivering multimedia data namely compressed video to fixed and mobile subscribers.

There is a strong movement towards convergence at multiple levels and particularly in view of potential efficiency gains obtained from integration. Therefore, convergence goes through integration of services, networks, systems, platforms and terminals. However, an efficient end-to-end content delivery through heterogeneous networks and terminals requires a serious and deep research in order to guarantee an acceptable QoS. Thus, SUIT, intends to broadcast and stream scalable and multiple descriptive visual contents in an optimal way through DVB-T/H along with IEEE802.16 (WiMAX) networks to homes and to extended home environments. The QoS (delay and bandwidth) is guaranteed by the intelligent multiplexer at the playout site and by requesting appropriate bit rates and spatio-temporal layers from the scalable [4] servers and live encoders. The proposed novel convergent system delivers multimedia data to users, including broadcasting, multicasting and unicasting under the big umbrella well known as all-IP. SUIT is an applied research project with a strong impact in the professional and consumer electronics markets. SUIT is the first project demonstrating real time scalable video broadcasting over broadband mobile networks. Due to its enormous challenges, SUIT has been developed by a consortium composed of 10 companies and research institutions. We have selected ISCE2007 to publish the first paper describing the SUIT architecture due to its proximity with and the consumer electronics.

REFERENCES


Antonio Navarro (S’89-M’97) graduated (five years first degree) in electrical engineering from Coimbra University, Portugal in 1989 and received the MSc and PhD degrees from the University of Coimbra, Portugal and the University of Newcastle, UK in 1993 and 1996, respectively. He is currently Professor at the Electronics and Telecommunications Engineering Department at Aveiro University, Portugal. In the 2nd semester of 2004, he was on sabbatical leave at University of Southern California-USA. His research interests are on information theory,
A Video Coprocessor for Mobile Multi Media Signal Processing

Benno Stabernack (Fraunhofer Institut Heinrich Hertz Berlin, Germany); Kai Wels (Fraunhofer Institut Heinrich Hertz Berlin, Germany); Heiko Hübert (Fraunhofer Institut Heinrich Hertz Berlin, Germany)

Mobile multimedia is becoming more and more popular using mobile phones or similar devices for the reception of multimedia content. One of the major drawbacks of current mobile multimedia systems is that content is delivered in a point-to-point configuration, which is not efficient in terms of required bandwidth of the network cell. Especially for the most important application, the broadcasting of audio/video (A/V) content, it has been shown that RTSP/RTP based point-to-point streaming systems are not applicable. To solve the problems of broadcasting A/V content for small portable devices, the new DVB-H standard has been developed, which addresses network aspects as well as the power consumption problems of the RF frontend in mobile devices. Based on IP datascast protocols an existing DVB-T infrastructure can be used for A/V broadcast transmission. According to the DVB-H specification transmitted audio data will be coded using AAC-HE, whereas the video content will be compressed using the H.264/AVC baseline profile. To support different scenarios in terms of coding quality, picture resolution and terminal capabilities the DVB-H standard defines profiles and levels with different data rates, see table I. Modern video coding standards like H.264/AVC require a high amount of computational power and memory space, which in general does not cause problems for platforms like PCs or Laptops. However, most mobile DVB-H terminals will be built around embedded processors, which underlie severe restrictions concerning computing power, memory usage and power consumption. It has been shown after intense application profiling of software based players, that decoding of H.264/AVC video data requires the largest amount of processing power in DVB-H reception, which is about 70%. Mobile devices have a powerful processor available, which might be sufficient for also overtaking the full task of H.264/AVC video decoding. An implementation of a DVB-H terminal based on a PDA (Dell Axim X50v), using the Intel XScale processor running at 624 MHz, has shown that a pure software implementation is possible [2]. This has the advantage of a short time to market, since the hardware is already available and the software development can be accomplished quickly. The disadvantage is the high power consumption, because nearly the maximum system frequency (approx. 600 MHz) is required for running the full DVB-H application. This leads to a short battery runtime. A fully hardware based implementation of the H.264/AVC video decoder leads to a very low power consumption and therewith long battery runtimes. However the development is very cost intensive and time consuming. Furthermore the design cannot be adapted to future variations of the standard. In a mixed hardware/software solution the control flow within the decoder is performed in a processor and arithmetical intensive parts are overtaken by specific coprocessors. In the best case, the coprocessors are configurable by the software in order to be adaptable to other video standards. This leads to a trade-off between flexibility, short time to market and battery runtime. In this paper we will show our approach to overcome the trade-off between performance, flexibility and chip area. By analyzing the H.264/AVC video coding standard in detail, we derived the most useful architecture on the basis of profiling results acquired by a specialized memory profiling tool. In order to prove the presented concepts, a companion chip architecture for embedded processors in DVB-H terminals has been developed incorporating a processor core with a specialized instruction set and a set of flexible coprocessors. The companion chip is a key component of a DVB-H terminal, which has been used for the currently running DVB-H field trial in Berlin, Germany. Due to similarity of the video coding parameters used by DVB-H and DMB, the described SOC can be used as well for DMB terminals. As an outlook we present the usage of the presented SOC architecture for the upcoming scalable extension of the H.264/MPEG-4 AVC video coding standard, which provides scalability in terms of picture resolution frame rate and PSNR for future broadcast applications.

Heiko Hübert received the Diploma degree in electrical engineering from the Technical University of Berlin, Germany in 2000. He has been a (visiting) researcher at Stanford University, USA, at the Royal Institute of Technology (KTH) Stockholm, Sweden and at the Rheinisch-Westfälische Technische Hochschule (RWTH) Aachen, Germany. In 1998 he worked as an intern for Ericsson Datacom Networks, Stockholm, Sweden. He is with the Fraunhofer Institute for Telecommunications, Heinrich-Hertz-Institut (HHI), Berlin, Germany, since 2002. His research focuses on embedded software and hardware for multimedia applications and on tool development for software analysis and profiling.
A comparison of DoJa and MIDP security models

Igor Valdin (St. Petersburg State Polytechnical University, Russia); Vladimir Rybkin (Motorola, Russia); Denis Vasilin (Saint-Petersburg Electrotechnical University, Russia)

Java ME is the most commonly used application platform for mobile devices. The Mobile Information Device Profile (MIDP) is today's most popular Java runtime environment for resource constrained devices. DoJa, being an analog of MIDP, was introduced by NTT DoCoMo in 2001 as a part of i-mode service provided to its customers. Millions of mobile phones all over the world have integrated MIDP or DoJa environment. There is a class of Java applications requiring an access to private user data. Such applications must be trusted and have to pass the authorization process according to the security policy implemented in each profile. This paper compares DoJa and MIDP security models. It contains an introduction to DoJa and MIDP architectures, overview of security mechanisms and traits of trusted applications implementation. Also the security enhancements expected in new MIDP and DoJa versions are analyzed.

Igor Valdin was born in 1984 in St. Petersburg, Russia. In 2001 he entered St. Petersburg State Polytechnic University, Faculty of Engineering Cybernetics. In 2007 he became IEEE Student Member. Since 2005 Igor works in Motorola St. Petersburg Software Center and participates in DoJa VM development for new Motorola platforms. He specializes in Java Application Manager and low-level graphics implementation.

Technological Approaches for Secure DATA Processing in Networked RFID Architectural Framework

Namje Park (ETRI, Korea); hoWon Kim (ETRI, Korea); Dongho Won (SKKU, Korea)

* Introduction Networked RFID service is defined as a special type of mobile service using RFID tag packaging object and RFID reader attached mobile RFID terminal. This paper is to provide an approach for ensuring RFID network security based on security engineering method. In order to ensure secure mobile RFID service network, we describes an approach which includes the concept of security state and security flow. Consequently we will propose the way to protect the personal information privacy effectively using security preference on this paper. The mobile RFID is a technology for developing a RFID reader embedded in a mobile terminal and providing various application services over wireless networks. Various security issues – Inter-domain security, privacy, authentication, E2E (End-to-End) security, and untraceability etc. - need to be addressed before the widespread use of mobile RFID. Model of mobile RFID service defines additional three entities and two relationships compared to that defined in RFID tag, RFID access network, RFID reader, relation between RFID tag and RFID reader, relation between RFID reader and application server. Generally, in mobile RFID application such as smart poster, Application Service Provider (ASP) has the ownership of RFID tags. Thus, mobile RFID users have to subscribe to both the ASP for these kinds of RFID services and mobile network operator for mobile communication service. Namely, there exist three potentially distrusted parties: user owned RFID reader, mobile network operator, and ASP. Accordingly, trust relationship among three parties must be established to realize secure mobile RFID service. Especially, when a RFID reader tries to read or change RFID service data stored in tag, the reader needs to get a tag access rights. Additionally, it is important that new tag access rights whenever some readers access a same tag must be different from the already accessed old one. (1) Mobile RFID Crypto Library : The Mobile RFID Crypto Library is the crypto library for efficient processing of the crypto algorithms and security protocols. It provides security mechanisms to the Mobile RFID Reader and it is targeted to the Mobile RFID middleware based on WPI platform. The Mobile RFID Crypto Library enables Mobile RFID service provider, wireless contents provider and information security industry to support information protection service on mobile RFID middleware platform with reasonable cost and in short period of time. (2) Secure Mobile RFID Middleware in phone : The Secure Mobile RFID Middleware is a system with which a user can acquire information on desired objects through mobile communication network by installing RFID reader module or chip onto the cellular phone to control RFID Reader, and by safely providing various computing functions and filtering functions onto tag recognized through the reader as well as ODS Query-related process functions to Mobile RFID Application system. Transplanting and extending security library for protecting mobile RFID Information in this system to support security application on the path of all data from RFID Reader to application server. (3) Secure Mobile RFID H/W Reader : Mobile RFID reader is a handheld 900MHz RFID reader and can be used as a peripheral of the mobile devices such as a smart phone and PDA. Its compatibility of EPC C1G2 and the capability of wireless communication provide many possible applications. (4) Mobile RFID Privacy Protection Service System : The RFID user Privacy management Service provides Mobile RFID users with information privacy protection service for personalized tag under Mobile RFID environment. When an RFID tag is personalized, RPS enables the owner of the tag to control his backend information connected with the tag such as product information, distribution information, owner's personal information and so on. (5) Secure Mobile RFID Application Portal Platform Systems : The SMAP(Secure Mobile RFID Middleware) is a secure service portal for various Mobile RFID application services. The service provider using SMAP can easily construct Mobile RFID application guaranteeing security and privacy protection * Conclusion The mobile Networked RFID technology is being actively researched and developed throughout the world and more efforts are made for the development of related service technologies. Though legal and institutional systems endeavor to protect privacy and encour-age protection technologies for the...
facilitation of services, the science and engineering world also has to develop proper technologies. Seemingly, there are and will be no perfect security / privacy protection technology. Technologies proposed in this paper, however, would contribute to the development of secure and reliable network RFID circumstances and the promotion of the mobile RFID market.

Friday, Jun 22

8:30 AM - 9:30 AM

Keynote 5

Room: Trinity 1-4

9:30 AM - 10:30 AM

Consumer Networks 4

Room: Trinity 1-3

*Use of fault tree analysis to improve residential gateway testing*

Sakkaravarthi Ramanathan (France Telelecom R&D, France); Chidung Lac (France Telecom R&D, France)

The residential gateway (RG), heart of the strategy of most Telcos, is a centralized intelligent interface between the operator's network and home network. It terminates all external access networks to the home and enables residential services to be delivered to the consumer, including both existing services and new ones yet to come. Besides a plethora of useful services, the growth of RG in market depends upon the reputation of its resilience (availability, reliability and security). This emphasizes a fault free design and efficient testing should be taken care before the launch of the product into the market. Even though the RG is tested conscientiously for its services before launch, some major failures are still prevailing in the customer premises. This paper uses fault tree analysis (FTA) to capture the failures that successfully pass through the testing phase and generate catastrophic effects in the field. With the help of standard requirements defined by specifications, we have identified the zones in which testing in the laboratory needs to be improved.

Chidung Lac is a senior researcher in the R&D Division of France Telecom, where he has managed during three years an R&D unit whose activities are focused on dependability and performance of access and transport networks. His current research interests include resilience of communication services. Dr. Lac earns a Ph.D. in Physical Sciences from the University of Paris XI (1987).

*A Dynamic GTS Allocation Algorithm in IEEE 802.15.4 for QoS guaranteed Real-time Applications*

JunKeun Song (Electronics and Telecommunications Research Institute, Korea); Jeong-dong Ryoo (ETRI, Korea); Sang Cheol Kim (Electronics and Telecommunications Research Institute, Korea); JinWon Kim (Electronics and Telecommunications Research Institute, Korea); Haeyong Kim (Electronics and Telecommunications Research Institute, Korea); PyeongSoo Mah (ETRI, Korea)

In this paper, we propose a novel QoS and real-time guaranteed algorithm in the IEEE 802.15.4 protocol. Recently, The IEEE 802.15.4 protocol is intended to serve a set of industrial, residential and medical applications with very low power consumption and cost. There are also many real-time applications in wireless personal area networks. IEEE 802.15.4 provides a Guaranteed Time Slots (GTSs) mechanism to allocate a specific duration within a superframe for time-critical transmissions. However, there are many weak points to support real-time applications; First, The active period of superframe is split into 16 time slots and one node has to use at least one whole GTS. Each node may partially use the slot due to the pre-fixed time slot durations. Therefore, using more GTSs, the amount of wasted bandwidth will be increased. Second, the maximum number of GTSs is limited. It means that only a few nodes can use GTSs. Third, the Pan Coordinator can change the superframe cycle to adapt some applications. But if two applications have to send periodic messages with different cycle, it can’t be handled. To address these problems, a new admission control algorithm is presented that enables a GTS to be used by more than two nodes at the same time for maximizing the utilization. The proposed algorithm also allocates the GTS dynamically to support real-time applications. We evaluated the delay bounds and the throughput of the proposed scheme compared with original protocol. We then showed that our proposal improves the bandwidth.
utilization and guarantees real-time features more strictly.

**A Study on the Realtime Message Report Procedures and Management Scheme for the Quality Guaranteed VoIP Services**

Hyun-Woo Lee (ETRI (Electronics and Telecommunications Research Institute), Korea); JinS Kim (ETRI, Korea); Won Ryu (ETRI, Korea)

This paper provides message report procedures and management schemes to guarantee QoS on the internet telephone. In order to establish and release for managing quality guaranteed VoIP services, we present 8 critical message procedures with RTCP-based packet structures BTs which are based on standard RTCP-XR format to perform better monitoring of quality factors such as jitter, delay, loss, etc. For the reporting quality parameters optimally during establishing call sessions of VoIP service, we design two critical management module blocks for call session and for quality reporting. To prove the proposed method, we apply of reporting procedure and management scheme based on case by case in various experiments. The experimental environment is composed as follows; VoIP-Quality Management server, SoftPhones, 1-port G/W, and Softswitch for managing an end-to-end speech quality over IP network. Overall, for the evaluation with scientific exactitude of quality factors, we examine by phone-call services through heterogeneous network systems: PSTN to SIP gateways. The experimental results confirm that realtime message report procedures and management schemes achieve a sufficient for reporting and managing of quality factors for the quality guaranteed realtime VoIP services.

**IMS session control architecture for supporting QoS in FMC**

Hyun-Jin Lee (Electronics and Telecommunications Research Institute, Korea); Hwa-Suk Kim (Electronics and Telecommunications Research Institute, Korea); Cho Kee Seong (Access Mediator Research Team, ETRI, Korea)

Currently IMS (IP Multimedia Subsystem) products are being appeared but most of them are based on 3GPP Release 5/6 and they are usually used exclusively for fixed network or mobile network. So they can control the call/session of mobile network only or they implement their own QoS (Quality of Service) control architecture for fixed network. In this paper, we propose the IMS session control architecture to support QoS concurrently in fixed and mobile network. We referenced the network architecture of ITU-T (International Telecommunication Union - Telecommunication Standardization Sector) NGN (Next Generation Network) framework, ETSI TISPAN (Telecoms and Internet converged Services and Protocols for Advanced Networks), and the standard interface Rx in 3GPP Release 7. We used push-pull mode mechanism to support QoS in mobile network and push mode mechanism in fixed network. In push-pull mode, the wireless network authorizes QoS, the UE (User Equipment) requests real bandwidth needed, and the network commits QoS. On the other hand, in push mode, the fixed network does all of them, i.e., authorizes QoS, reserves the bandwidth, and commits QoS. We put the QoS control logic of fixed/mobile network in IMS server, especially in P-CSCF (Proxy Call Session Control Function) and loaded the Rx protocol, which is an application of Diameter Base Protocol. The IMS server consists of S-CSCF (Serving CSCF), I-CSCF (Interrogating CSCF), and P-CSCF. We also implemented the access/transport network simulator for the counterpart of P-CSCF and put the QoS execution logic in it. Lastly we stuck the fixed/mobile integrated UE simulator to the access/transport network simulator to prove the QoS supported IMS session control architecture. We detailed the signaling flow for fixed/mobile UE to/from fixed/mobile UE and fixed/mobile UE to/from non-IMS UE. Non-IMS UE means the UE in the network that is not based on IMS, i.e., Soft Switch based network or general SIP server based network. We tested with Voice/Video Call service and identified that the proposed IMS session control architecture successfully supports QoS concurrently both type of UEs.

He received the B.S. and M.S. in electronics from Kyungpook National University, Daegu, Korea. He joined the ETRI (Electronics and Telecommunications Research Institute) in 1999. Since then, he has been engaged in the research and development of Mobile Switch system based on ATM, Optical Internet, and Access Mediator system. He is currently working in fields of Broadband Convergence Network Control Technology in ETRI. His interests include the areas of next generation network control system, IP Multimedia Subsystem, and Converged Services.

**Home Entertainment 1**

Room: Trinity 4

**TV-Anytime Metadata Authoring Tool for Personalized Broadcasting Services**

Yang Seung-Jun (ETRI, Korea); Jungwon Kang (ETRI, Korea); Dong-San Jun (ETRI, Korea); Minje Kim (Electronics and Telecommunications Research Institute, Korea); Han-kyu Lee (ETRI, Slovakia)

It is expected that, in the near future, multimedia data service providers will provide various kinds of data to satisfy the demand of audience in digital broadcasting environment. The TV-Anytime Forum specifies a set of metadata to be used for efficient access and browsing of broadcasting content in a personalized way. The metadata include
plenty of information about the content that may be utilized for the electronic program guide, content-based search and browsing, and adaptation of the content to diverse terminal and network conditions. A couple of practical schemes that use the metadata in digital broadcasting can be found in preexistence papers. In spite of a useful service based on TV-anytime metadata, the metadata authoring still remains as a harassing and time-consuming task. Although some kinds of metadata are extractable automatically, there are still needs of human intervention for refining the results (esp., in subjective terms) and annotating manually the fields that are difficult to extract in any automated means. Therefore, an efficient authoring tool is required. In this paper, we present a design and implementation of a TV-Anytime metadata authoring tool to provide personalized broadcasting services. For easily metadata authoring, the proposed metadata authoring provides the following key functionalities: metadata visualization, media access, and semiautomatic method for editing segment related metadata. The proposed TV-Anytime metadata authoring tool is to provide users visual and intuitive environment for authoring content descriptive metadata. We implemented the metadata authoring tool about TVA-1 and package metadata authoring tool among TVA-2. The implemented authoring tool provides various kinds of functionality as follows: - Metadata loading: load existing metadata to the memory device after parsing and validation - VCR function: support basic VCR functions including play, pause, fast-forward, and random access - Segment extraction: be equipped with some automation tools for authoring segment metadata - Visualization: visualize the content of metadata in an effective way and support editing in the visualized environment - Audit: audit user interactions real-time - Save: save results into a valid XML (eXtensible Markup Language) document The proposed visual metadata authoring tool provides the convenience of user at authoring step. We expect that this authoring tool can be applied to various personalized broadcasting services such as DTV, DCATV, DMB and so on.

News package service based on TV-Anytime metadata gathered from RSS ★

Hee Kyung Lee (Electronics and Telecommunications Research Institute, Korea); Hui Yong Kim (ETRI, Korea); Han-kyu Lee (ETRI, Slovakia)

Recently, due to the diversification and digitalization of the broadcasting and the convergence of the broadcasting and telecommunication, there have been rapid progresses in the broadcasting environments so that users can utilize the bi-directional return path. According to the tendency, TV-Anytime Forum has already standardized the system and method for bi-directional metadata service. To provide attractive bi-directional metadata service, the service provider needs to gather a large number of contents from contents providers through individual contracts, and then author metadata for all the gathered contents. In order to reduce service provider’s load and easily provide rapidly updated contents, we propose to make use of RSS (Rich Site Summary) service. RSS is simple data format based on XML (Extensible Markup Language) which is utilized for community of information via web site. Additionally, member elements of RSS are very similar to those of TV-Anytime metadata, so they can be easily transformed to the TV-Anytime metadata. RSS service collects frequently updated contents such as news, weather, engagement announcement, and etc. from already contracted content providers. And it manages them, and informs most recent contents using RSS to users. Taking advantage of RSS data and RSS contents, in this paper, we present news package service based on TV-Anytime metadata gathered from RSS service. The proposed news package service system consists of five functional units, i.e. RSS service provider, RSS collector, RSS manager, bi-directional metadata service, and RSS consumption tool. RSS service provider provides up-to-date news contents using RSS. RSS collector gathers RSS data using genre and keyword set by RSS manager. And it extracts AV contents’ location and news’ script from RSS data. And then, news package are generated from them. RSS manager sets genre and keyword for RSS collector. In addition, it classifies news packages as hot news, popular news, and special news. According to the classification, news packages are differently provided to users. Bi-directional metadata service receives user request including retrieval parameters for news packages, and then it retrieves contents & metadata DB using the parameter to find proper news packages. The list of retrieved packages is sent to the RSS consumption tool. RSS consumption tool retrieves, selects, and consumes news package. In brief, the proposed news package service gathers news contents from pre-existing RSS news service, extracts TV-Anytime metadata from RSS data, and generates news package. Consequently, this method greatly mitigates the load of bi-directional metadata service provider and makes it easy to provide rapidly updated contents. By these effects, bi-directional metadata service provider could easily enter into the field.

Personalized Data Broadcasting Service based on TV-Anytime metadata ★

Ji Hoon Choi (ETRI, Korea); Joo Myoung Seok (Electronics and Telecommunications Research Institute, Korea); Seong Yong Lim (ETRI, Korea); Hyun-Cheol Kim (ETRI, Korea); Han-kyu Lee (ETRI, Slovakia); Jin-Woo Hong (ETRI, Korea)

As the broadcasting environment is evolving into a digital and digital broadcasting industry is activated, a trend of digital broadcasting research has been changed from a simple watching broadcasting service to a user custom broadcasting service because of several reasons. Data broadcasting service is one of simple watching broadcasting services has became to play an important role to provide users with high-definition AV, high quality audio like CD and additional multimedia data such as text, still images, moving pictures and so on. Also, it enables
to provide viewers with interactive services through a bidirectional channel or a return channel. Furthermore, although digital broadcasting environment might be better as the transmission rate of broadcasting channel is expanded and broadcasting channel/program numbers are increased, researches for digital broadcasting consumption environment have not been activated. For example, it must be difficult to users to choose favorite channel or program in multi-channel broadcasting environment without additional information of broadcasting channels and programs although digital broadcasting announcement standards exit, such as PSIP(Program and System Information Protocol) and DVB(Digital Video Broadcasting)/OCAP(OpenCable Application Platform)-SI(System Information) In order to solve above mentioned problem, TV-Anytime metadata standard considering various digital broadcasting consumption environment has been applied to digital data broadcasting. As result of that, research of personalized data broadcasting service based on TV-Anytime metadata will be going on actively. In this paper, we suggest personalized data broadcasting service model and method to provide easily user with broadcasting contents that viewer wanted using by private profiles, program summary information and program index information. And then, we explain example and architecture of TV-Anytime metadata for personalized data broadcasting service. Finally, we show a result of the personalized data broadcasting service model such as My Guide, My TV, Search TV and My History service. First of all, My Guide service using by TV-Anytime package metadata and service information metadata provides user with broadcasting program information on type of mosaic EPG(Electronic Program Guide) according to genre, theme, user preference keyword such as drama, sports, news, world cup games, top-rated program 10 and so on. Secondly, My TV service provides user with detail information and segment information of recorded programs as well as ToC(Table of Content) information, highlight information, bookmark information. And then, it enables to provide user with trick mode service using by various type of segment information. Thirdly, Search TV service provide user with result of program search using by keyword information, program detail information, character information, genre information defined in TV-Anytime. Finally, user's favorite channels or programs information according to user's watching history are managed automatically in user history service. If user history information is send to broadcasting provider and service provider, it will be useful information of a program rating. Also it will be used to support more personalized data broadcasting service too. In the future, we have plan to maximize an application of TV-Anytime metadata on digital broadcasting environment and to research user creation broadcasting service on convergence digital broadcasting environment.

Ji Hoon Choi received the B.S and the M.S degrees in electronic engineering from Kyunghee University, Suwon, Korea, in 1999 and in 2001, respectively. In 2001, he joined the Data Broadcasting Research Group in ETRI(Electronics and Telecommunications Research Institute, Deajeon, Korea, where he have been working on the interactive data broadcasting system. His research interests are the data broadcasting and network QoS.

**Mobile Computing & Communication 2**

Room: Trinity 6-8

**A macroblock-level analysis on the dynamic behaviour of an H.264 decoder**

Ralf Schreier (Vienna University of Technology, Austria); Florian Seitner (Vienna University of Technology, Austria); Michael Bleyer (Vienna University of Technology, Austria); Margrit Gelautz (Vienna University of Technology, Austria)

This work targets the optimization of multi-processor H.264 decoder implementations. We have extended the simulator of a multi-core VLIW media processor to enable cycle-accurate function profiling on a sub-macroblock level which allows measuring the effects of coding modes on the computational complexity with very fine granularity. This knowledge helps the system designer to optimize the system performance and memory sizes to reduce system costs.

Florian H. Seitner received his diploma degree (M.S.) in Computer Science from the Vienna University of Technology in 2006. Since September 2006 he is employed as a project assistant at the Vienna University of Technology where he works on his Ph.D. degree. He is currently doing research on modeling of video coding algorithms on embedded multi-core architectures.

**Novel Multi-User MIMO Scheme Based on Successive Interference Cancellation**

Heesoo Lee (ETRI, Korea); Hyojin Lee (ETRI, Korea); Hyun Kyu Chung (ETRI, Korea)

The multiple input multiple output communications structures have received considerable attention recently as they could provide very high data-rate communication over wireless channels without increasing the total transmit power. In particular, a lot of researches have dealt with multiple codeword (MCW) MIMO system. For example, there are per antenna rate control (PARC), selective per stream rate control (S-PSRC), and so on. In MCW MIMO system, each antenna transmits separately encoded codeword that is called a stream. Each data stream forms an independent coding block and thus, the SINR estimation and rate control can be done on a stream by stream basis. The use of independently coded streams also has the advantage in performance by allowing reliable
successive interference cancellation (SIC) based on post-decoding symbols at the receiver, which is not permitted in the conventional single-stream transmission, that is, single codeword (SCW) MIMO. When the receiver has SIC capability, the signal to interference noise ratio (SINR) can be increased, and hence the transmitter sends data with higher rate. An information theoretic result says that the PARC principle can achieve the Shannon capacity limit for an open loop multiple antenna system if separately encoded data streams are transmitted from each antenna with equal power but possibly with different data rates, and if the receiver consists of a space time MMSE linear filter followed by interference cancellation based on post-decoding symbols. Per unitary basis stream user and rate control (PU2RC) is a different kind of MCW MIMO system from PARC or S-PSRC. PU2RC is multi-user based transmission scheme. While all streams (codewords) are aimed at one specific user in PARC and S-PSRC, PU2RC scheme transmits multiple data streams to multiple users simultaneously to utilize the multiuser diversity in the spatial domain. In this multiuser scheme, a maximum performance can be achieved by allocating each stream to a “good” user equipment (UE) with the aid of SINR’s of the data streams reported by all UE’s. In addition to this difference, PU2RC uses the precoding MIMO technique with a predetermined set of multiple precoding matrices. A set of unitary matrices including the identity matrix was suggested as precoding matrices. In general, PU2RC can achieve more multi-user diversity gain than PARC or S-PSRC. However, PU2RC cannot exploit SIC gain to increase transmission rate, although the receiver itself may employ SIC as in PARC. When the total number of users in the network is large, PARC has a larger system throughput than PU2RC due to full use of SIC for increasing transmission rate, whereas as the number of users increases PU2RC has a larger capacity than PARC due to exploiting full multi-user diversity. In this paper, we propose two novel multi-user MIMO schemes, which can make full use of SIC gain to increase transmission rate and also multi-user diversity gain. The first is successive interference cancellation (SIC) based Per User and Stream Rate Control (S-PUSRC), and the second is a modified scheme of S-PUSRC. S-PUSRC is compatible with any MIMO transmission schemes employing independently coded multiple streams and a predetermined set of precoding matrices or vectors. In the multiuser MIMO scheme PU2RC, the UE reports back the SINR’s for each data stream for a given unitary precoding matrix and an index to the unitary matrix. The SINR’s for data streams can be obtained after estimating MIMO channel coefficients using pilot symbols. Although the receiver itself may employ SIC as in PARC when more than one data streams are allocated to the UE, the estimation of the SINR’s that are reported back by the UE is solely based on the detection of the received signals without incorporating SIC process and information regarding the stream ordering for decoding and cancellation is not provided to the BS. In S-PUSRC, the receiver is assumed to perform SIC based on decoding symbols, and the feedback information consists of the decoding order and the post-detection SINR’s for each data stream estimated under the assumption of perfect cancellation of preceding streams in SIC. In more detail, the decoding order and the post-detection SINR’s are obtained through following processes. First, post-detection SINR’s for each data stream are calculated using the estimated channel coefficients, and the data stream with the largest post-detection SINR is chosen to be at the top of the decoding order list. Assumming that the decoding and cancellation process at the receiver exactly removes the interference due to the selected stream from the received signals, post-detection SINR’s are re-estimated for remaining data streams. Again, the stream with the largest post-detection SINR is selected as the next stream in the decoding order. Repeating this process, the decoding order and its associated post-detection SNR’s are obtained. To obtain the benefit of the feedback of the decoding order and SIC-based SINR’s, in combination with an appropriate multiuser scheduling algorithm, is that multiuser diversity can be fully exploited in stream allocation and simultaneously, the SINR gain due to the SIC process at the receiver when more than one data streams are allocated to one UE brings about improvement in system performance. Once the stream allocation is decided, the modulation and coding scheme (MCS) for the allocated data streams can be determined using the corresponding reported SINR’s. The BS decides on which data stream is to be allocated to which user by taking into account the feedback information collected from all active users but under the following two constraints. 1. One data stream cannot be allocated to more than one user. 2. The i-th data stream can be allocated to a certain user only when the data streams that precede the i-th stream in the decoding order list of the user have been allocated to the same user. When the total number of available users is small PARC has a larger capacity than PU2RC due to its capability of SIC whereas as the number of users increases PU2RC has a larger capacity than PARC due to increasing user diversity. The newly proposed scheme S-PUSRC has the largest capacity regardless of the number of users. When the number of users is small (< 3), S-PUSRC shows much similar capacities with the single-user scheme PARC. However, as the number of users increases, the gain from the multiuser scheduling exploiting multiuser diversity begins to play a role and the capacity gain over PARC widens. The second proposed scheme (modified S-PUSRC) reduces channel quality information (CQI) feedback overhead with slight throughput degradation in comparison with S-PUSRC. In modified S-PUSRC, the feedback period of decoding order is quite longer than that of SINR. Simulation results show that the modified S-PUSRC scheme performs better than PARC and PU2RC in all cases while the modified S-PUSRC scheme has the same amount of feedback overhead as PARC and PU2RC.

Heesoo Lee received the B.S., M.S., and Ph.D. degrees in management science and industrial engineering from KAIST. Currently, he is a senior researcher at Electronics and Telecommunications Research Institute (ETRI) in Korea. His research interests are in the areas of next generation wireless communication, MIMO, link adaptive...
transmission, and multihop relay.

**Power Consumption Reduction in Java-enabled, Battery-powered Handheld Devices through Memory Compression **

Mayumi Kato (The University of Texas at San Antonio, USA)

Mobile/wireless services have been developed using Java technology on battery-powered handheld devices. These services such as games, web browsers, and audio/video players are easily accessible but consume a huge amount of power. The handheld devices typically last for less than 6 hours. Therefore, a technique for low-power consumption is needed. This paper presents a runtime power control mechanism to reduce power consumption in Java-enabled, battery-powered handheld devices. The runtime power control is integrated into a Java runtime environment (JREPC) using a hardware/software codesign paradigm. The technique uses compression to minimize memory demands and employs a memory partition technique to map the demands on memory banks and power off the unused banks. JREPC is studied and evaluated with benchmark applications. Simulation results show that an in-memory compression reduces more than 50% of memory demands. Only half of the memory is activated and power consumption is reduced by 50%. The time overhead due to the in-memory compression is negligible using a hardware compressor and a hardware decompressor. Thus, the in-memory compression with the memory partition technique is one of the effective methods to achieve low-power consumption in the Java-enabled, battery-powered handheld devices.

B.A. (Gunma Prefectural Women's University 1988) is a Ph.D. Candidate, Department of Computer Science at The University of Texas at San Antonio. Her research interests include: Computer Systems and Architecture, Operating System, Programming Languages.

**A Novel approach to parallel channel transmission in Digital Terrestrial and Handheld Television **

Scott Linfoot (De Montfort University, United Kingdom)

The radio spectrum is fast becoming a scarce resource especially in light of all the many applications that utilize the higher Radio Frequencies (RF). It is, therefore, important that this resource is carefully managed to avoid waste. To help optimize the RF spectrum, in 1997, The United Kingdom launched the worlds first commercial digital terrestrial television service with the ultimate aim to switch off the analogue services between 2008 and 2012 thus potentially freeing up a large number of RF channels for either more television services or for other applications. The UKs digital television infrastructure is compliant with the ETSI EN 300 744 standard. Because the UK was eager to release the analogue bandwidth as soon as possible (meaning the government had to start to migrate quickly), the UK adopted the 2K version of DVB-T due to commercial hardware restrictions at the time (the 8K system was too expensive for use in consumer devices). Those countries moving over to digital television recently have tended to go straight to the 8K system because the technology is now commercially viable and because of all the advantages the 8K system has to offer. Recently, focus has shifted from research into terrestrial television for static receivers as found in the home to mobile receivers as found in such consumer devices such as mobile phones and portable hand held televisions. The infrastructure, although primarily based on DVB-T has been given the acronym DVB-H (DVB-Handheld). The main consideration with such a technology is that of power consumption. One way to lower the consumption is to reduce the complexity of the systems used such that less powerful cores can be used. OFDM is the underlying technology of the DVB-T and DVB-H standards and comprises of an Inverse Fast Fourier Transform (IFFT) at the transmitter (and an FFT at the receiver) which performs the frequency division multiplex (unlike wideband communications which tend to use time division multiplex). It is the size of these IFFTs and FFTs that determines whether the system is in the 2K or 8K mode. The implementation of such large FFT cores was the main cause for delay for implementing OFDM in consumer devices although OFDM was used in military applications long before this time as there are minimal budget constraints in military hardware. In 1997, the 2K core was just becoming commercially viable with a Set-Top Box (STB) costing the consumer in the region of £200-300. Today, most STBs are dual core (implementing both systems) and can cost as little as £30 for a basic system. If the technology could be further simplified, then it would be of significant benefit to the manufacturing community as it would contribute to a drop in production costs impact on the consumer market. Even with its inherent advantages, there is a drawback to OFDM being the inflexibility of the system. With OFDM, there are a small number of parameters that can be changed to suit the channel but the choice has to apply across a whole OFDM symbol — it is not possible to code different parts of the symbol in different ways as it is in the Japanese Integrated Services Digital Broadcast (ISDB) standard. With all this in mind, this paper will propose an alternative to using OFDM in DVB-T and DVB-H which employs a true time-frequency division multiplex using wavelets which will provide a more flexible environment that can be tailored to suit signal and channel conditions and of lower financial and computational cost. Investigations will be presented that show that this technique (entitled Orthogonal Wavelet Division Multiplex) could provide a highly flexible, channel optimized modulation technique based on time-frequency multiplexing and this paper presents the preliminary phase of investigations into this new technique.

Dr Linfoot was born in the UK in 1976. In 1998, he graduated from teh univeristy of Redding, UK with a BEng
degree in Electronic Engineering and in 2003, He graduated his PhD in the field of equalisation of Digital Terrestrial Television. He has been an academic since 2001 and has recently moved to De Montfort University as a Senior Lecturer where he is a member of the Wireless Multimedia and Signal Processing Research group. His research interests are channel optimisation and signal processing of terrestrial channels as well as audio engineering.

10:45 AM - 12:00 PM

Consumer Networks 5

Room: Trinity 1-3

Performance Evaluation of AES Algorithm on Various Development Platforms
Chirag Parikh (University of Texas at San Antonio, USA); Parimal Patel (University of Texas at San Antonio, USA)

With the rapid growth of low-cost, low-power handheld devices with wireless capability, more and more devices connect to wireless networks. This warrants secured data exchange and authentication to take place during the connection. 802.11i security scheme is becoming a de-facto standard of future WLAN, which uses AES algorithm in both encryption and authenticity of payload data. Cryptographic transformations of AES are computationally intensive, consuming significant power. Most of the published AES implementations target high-end products with multi-gigabit throughputs and have no regards for power consumption. Low-end products, including handheld devices like PDA rarely need more than 200 Mbps data transfer rate which necessitates alternative implementation of AES algorithm, with the aim of increasing the handsets’ battery life. In this paper we study and compare the outcome obtained from implementing AES algorithm on various platforms. We present results obtained through profiling the implemented algorithm on platforms like FPGA, Desktop PC (Microsoft Visual Studio) and handheld device (UIQ SDK for Symbian OS along with Metrowerks CodeWarrior). Our comparison shows that the 32-bit hardware implementation of AES on FPGA supports the required throughput and consumes less power allowing fully charged PDA to remain connected to WLAN for a longer duration.

Advanced Scheme to Reduce IPTV Channel Zapping Time
Jieun Lee (Korea Telecom, Korea); Geonbok Lee (Korea Telecom, Korea); Seunghak Seok (Korea Telecom, Korea); Byunghoek Chung (Korea Telecom, Korea)

The convergence is going on between broadcasting and IP communication. IPTV, the representative converged service, just has been spread throughout the world. For the success of IPTV service, it is a key feature to satisfy QoS (Quality of Services) of total IPTV solution. Intolerable channel zapping time will especially result in the dissatisfaction with IPTV service. The service delay during channel switch will be increased by geometric progression as the number of IPTV subscribers is increased. A recent survey shows that channel zapping time more than 1 or 2 seconds will lead to the failure of IPTV business. A related work reduces IPTV channel zapping time using a home gateway to take a role of IGMP-proxy. When a subscriber changes the current channel to a new one, a home gateway joins not only the multicast group for the new channel but also those for adjacent channels of the new channel in advance. Since then the home gateway can receive multicast traffic of both the new channel and the adjacent ones. Afterwards, if the subscriber changes IPTV channel to one of the adjacent channels, the home gateway can forward the multicast stream immediately. However, the related work considers the case that a subscriber selects an adjacent channel of the current one. If a subscriber uses hot-key on the remote controller or uses number button directly, the proposal doesn’t work any more. Additionally it is too inefficient that each subscriber must have his/her home gateway. In this paper, we propose a solution for the two problems using rating server. Firstly, to remove an IGMP-proxy home gateway, a set-top box behaves as the IGMP-proxy home gateway in the related work. A set-top box, instead of a home gateway, manages lists of both current channel and the adjacent ones. If channel switch is occurred, a set-top box joins multicast groups for a new channel and the adjacent ones. And then it leaves the multicast groups for the previous unnecessary channels and updates the managed channel lists. Secondly, to improve probability for expected channel, a set-top box obtains an expected channel list by sending a query message to rating server and joins the multicast groups for the obtained channels. Rating server collects the audience rating, generates and manages statistical data per each subscriber. For the rating data, all set-top boxes send rating data packets when each channel switch is occurred. When query message is received from a set-top box, rating server calculates expected channel from statistics of audience rating and sends back the response of the query with the expected channel list. In conclusion, our proposal can reduce a channel zapping time for IPTV service without a home gateway for each subscriber.

http://edas.info/showProgram.php?_qf__showProgram=&c=5136&form...bio%5D=1&program_view=everybody&action=Show+conference+program
**Design and Implementation of QoS guaranteed Bridge System for High Speed PLC and UWB**

*Dong-Hwan Park* (Electronics and Telecommunications Research Institute, Korea); *Tai Yeon Ku* (ETRI, Korea)

I. INTRODUCTION
Due to recent wireless technology developments in digital consumer electronics, Ultra Wideband (UWB) is becoming more attractive for low cost personal communication applications. As more and more mobile multimedia devices require large amount of bandwidth and may need accurate synchronization for real-time video transfer, UWB systems are naturally going to be utilized for the multimedia streaming of high-bandwidth contents. However, multimedia consumer electronics device using UWB cannot use and move at coverage of whole house area because of UWB’s short distance communication characteristics of high data rate. The latest developed high speed PLC technology can guarantee QoS of multimedia streaming in home network. If we use the high speed PLC to link with UWB, we do not need additional wiring for high bandwidth multimedia streaming and UWB’s signal extension. Accordingly, this paper proposes the high speed PLC and UWB bridge system to extend the UWB’s coverage to whole house area using high-speed PLC guaranteeing QoS. To guarantee uniform QoS mechanism in home network, this paper proposes the UWB-PLC bridge system adopt UPnP QoS architecture. II. HIGH SPEED PLC AND UWB BRIDGE SYSTEM

The proposed bridge architecture consists of 4 subsystems: HPAV QoS Subsystem, WiNET QoS Subsystem, Traffic Management Subsystem, and Bridge QoS Subsystem. Each subsystems of this bridge system are designed to minimize the interaction of each other. In this bridge system, we use the HomePlug AV (HPAV) as QoS guaranteed high speed PLC protocol, and the WiNET protocol as IP supported UWB protocol. UPnP QoS Device service is responsible for managing this bridge system’s QoS related resources for the requested traffic and returning the state of the bridge system. Adaptive QoS Manager manages an interaction between UPnP QoS Device Service and bridge’s QoS subsystems, and Universal QoS Transformer transforms TSPEC (Traffic Specification) to the Universal QoS description using XSLT. Universal QoS can minimize the dependency of the specified QoS parameter description such as TSPEC. HPAV QoS Subsystem has HPAV QoS Mapper, HPAV Link Monitor, and HPAV Connection Manager. HPAV QoS Subsystem performs the connection management of HPAV, the transformation of Universal QoS to HPAV’s CSPEC (Connection Specification), and the monitor of HPAV’s links status. WiNET QoS Subsystem is similar to HPAV QoS Subsystem. WiNET QoS Subsystem also has WiNET QoS Mapper, WiNET Link Monitor, and WiNET Connection Manager. Traffic Management Subsystem has two modules: Connection Table and Traffic Manager. The Connection Table contains information of all bridge’s connection. The Traffic Manager collect, store, update and delete the each connection’s information. When Classifier receives some frames, Classifier look up Connection Table’s information to find frame’s destination. Connection Table has classifying information: source IP address, destination IP address, source port number, destination port number, and protocol type. Additionally Connection Table has HPAN and WiNET’s connection Id for UPnP traffic. Bridge QoS Subsystem mainly divided by Control Plane and Data Plane. Local Resource Manager receives Universal QoS and traffic request information, and adjusts the bridge’s local resource using mapping on Local QoS Policy. Access Control determines the availability of QoS guarantee for the requested flow by UPnP QoS Manager. Classifier receives packets from HPAV and WiNET and looks up the Connection Table to find the destination of received packet. To find the packet’s destination, Classifier uses the HPAV and WiNET’s connection Id, source and destination IP address, source and destination MAC address, port number, protocol number, priority and input interface type. III. IMPLEMENTATION AND CONCLUSION

To verify the practicality of the proposed bridge system, a prototype of the proposed bridge has been implemented using INT6000 HomePlug AV solution of Intellon and WiNET prototype provided by Intel. The proposed high speed PLC and UWB bridge system for guarantee QoS can extend a limited UWB’s signal boundary to whole house area. Dong-Hwan Park received the B.S. and M.S. degrees in electronics engineering from Kyungpook National University, Korea in 1999 and 2001 respectively. Since 2001, he has been a research engineer of Home Network Middleware Research Team at Electronics and Telecommunications Research Institute (ETRI), where he develops the home network middleware and data broadcasting middleware. His research interests in home network middleware, wireless sensor network and pervasive computing.

**Fragmentation performance enhancement over network processor based MIPv4 home agent**

*Wook Kim* (Korea University, Korea); *Woojin Park* (Korea University, Korea)

Network processors are emerging as a programmable alternative to the traditional ASIC-based solutions in scaling up the processing of network services. In general, network processors enable us to deploy network services through software while maintaining high throughput and low latency. In this paper, we consider a home agent based on network processors. MIPv4 provides IP mobility to users. When a node moves to a foreign network, the mobile node registers its CoA (Care-of-Address) to the home agent by sending a binding update message. In order to route the packets to the mobile node that is on a foreign link, the home agent has to encapsulate the original packet and tunnel the packet to the CoA. In this situation, we should notice that the encapsulated packet can cause fragmentation. Current home agent which is based on network processors performs fragmentation in...
control plane just like other exception handlings. The home agent which is based on network processors performs fragmentation in control plane as an exception handling. If the encapsulated packets exceed MTU (Maximum Transfer Unit) size, the forwarding rate will be decreased rapidly due to processing overhead of the control plane. Therefore, more aggressive fragmentation techniques are required not to lower in forwarding performance. In this paper, we implement and evaluate a home agent that performs fragmentation in data plane, using IXP2800 network processors. In detail, we describe the overall architecture of a home agent, the procedure of the packet processing, and the key packet processing techniques. We evaluate and validate our system through internal and external benchmarks. Through internal benchmarks (Intel IXP Workbench simulator), it is found that the control plane used for packet fragmentation lowers the packet forwarding rate drastically. And this problem could be resolved by performing the fragmentation the data plane. By external benchmarks (SmartBits packet generator), we evaluate the performance characteristics of a home agent which supports the fragmentation in the data plane. Our experimental results and architecture can contribute to the design and implementation of network services over network processors.

**Design and Implementation of a Headend Cable Modem and a User Terminal supporting Multiple Downstream Channel Bonding**

Woongshik You (ETRI, Korea); Dong-Joon Choi (ETRI, Korea); Ohyung Kwon (Digital Broadcasting Research Division, Broadcasting System Group, ETRI, Korea); Soo-In Lee (Electronics and Telecommunications Research Institute, Korea)

In this paper, we present efficient designs and implementations of a headend cable modem and a user terminal supporting also proposed DOCSIS channel bonding scheme. The presented systems can enable various communication-broadcasting converged services over HFC network, such as HD IP-TV. The proposed DOCSIS channel bonding combines multiple downstream channels to provide greater aggregate bandwidth up to 400Mbps for a user. We also present the test and demonstration results showing the advantages of the proposed systems and DOCSIS channel bonding scheme. The proposed designs and implementations in this paper can be used to develop advanced headend cable modem and user terminal which provide hundreds Mbps transmission speed through DOCSIS channel bonding, and enable new converged services over HFC network.

**IPv6 anycast routing aware of a service flow**

Yoohwa Kang (Electronics and Telecommunications Research Institute, Korea)

The paper is related to the IPv6 anycast routing method for guaranteeing the continuity of the service flow. The terminal adds and deletes the unicast address mapped the anycast address in a anycast cache, and it unit looks up the anycast cache and requests the unicast routing by generating routing header. A server informs the unicast address about the anycast address of a terminal. When the server receives the unicast address, it deletes unicast address in the routing header received from the terminal and it changes into the anycast address. Therefore, a transmission is possible with one anycast server which is fixed by using the unicast routing between the terminal and server. By transferring the anycast address by using the routing header the anycast routing is comprised of an application and the continued service flow operates.

**ubiHome: An Infrastructure for Ubiquitous Home Network Services**

Young-Guk Ha (Electronics and Telecommunications Research Institute, Korea); JooChan Sohn (ETRI, Korea); Young-Jo Cho (ETRI, Korea)

Because of all the innovations in computer communications and digital consumer device technologies, people have had great interest in researching home networking systems. Most of which are mainly focused on home automation controls and audio/video streams or data exchanges between consumer devices. Leveraging the advantages of communication networks, such a system allows its user to control home climate, monitor home security and schedule a vacuuming robot with a TV remote controller while letting the user to watch VOD (Video On Demand) through the TV screen. Conventional research and development issues in this area have been related to communication media for home networking such as IEEE1394, PLC (Power Line Communication), Zigbee, WLAN (Wireless Local Area Network), IrDA (Infrared Data Association); middleware for home network systems such as HAVI (Home Audio/Video Interoperability), LonWorks, UPnP (Universal Plug and Play), Jini; and home network gateway platforms such as OSGi (Open Service Gateway Initiative). In the recent years, motivated by the emergence of ubiquitous computing technology as the next generation computing paradigm, a new class of home networks called “ubiquitous home networks” was introduced. Since then, some relevant projects have been researched and developed. Some examples are Aware Home, House_n, Home Sweet Ambient Home, EasyLiving and SOCAM (Service-Oriented Context-Aware Middleware). Ubiquitous home networks are actually home networks incorporating ubiquitous computing features. That is, conventional consumer devices can always communicate not only with each other but with ubiquitous computing resources such as wireless sensors, mobile/embedded computers and various information contents. And users can be provided with the service they need, anytime and anywhere in the ubiquitous home network environments. There are requirements to be met for
such a vision of ubiquitous home network services. One of the essential requirements is that service applications must provide services based on the awareness of current service contexts and environments. Here is a typical example for ubiquitous home network services, a home brightness control service. This service can control indoor brightness according to the current service environments such as user’s current location and available service devices nearby. The service application raises the window blind if the user is in the living room, or turn on the room lamp if the user is in the bedroom through a PLC controller. Consider a different service environment which has lightings with infrared switches, and a WLAN-enabled mobile consumer robot which is equipped with an infrared remote controller. Even in this case, the service application can also control indoor brightness by moving the consumer robot toward the lighting switch, and then using the infrared controller on the robot. Currently, existing approaches to the ubiquitous home network services are mainly based on proprietary service applications programmed for some specific service environments. For instance, Java applications for a specific dining room environment are used in SOCAM; and .NET applications for accessing specific sensors, lights and audio/video devices are used in EasyLiving. Surely, they can provide users with some level of ubiquitous services in the experimental home network environments. However, real service environments are dynamic, ad hoc and heterogeneous as presented in the above service example. To provide ubiquitous home network services effectively in the real service environments, service applications need to be automatically interoperable with sensors, consumer devices and other computing resources in the current service environments, rather than statically pre-programmed for specific environments. In this paper, we present design and implementation of an effective infrastructure for ubiquitous home network services called ubiHome. It enables automated integration of consumer devices, wireless sensors and other ubiquitous computing resources in home network environments by the use of Semantic Web Services, which is a state of the art Web technology. At first, we implement Web services for consumer devices, ubiquitous sensors and information contents as a unified interface method for accessing them. Then, we describe knowledge about capabilities and interfaces of the Web services in OWL-S (Web Ontology Language for Services), the semantic description language for Web services. And we register the knowledge to the service knowledge registry so that a home service agent can automatically discover the required service knowledge to collect context information and compose a feasible service process for the current environments. Finally, the service agent provides the service by automatically interacting with consumer devices and ubiquitous sensors through the use of SOAP (Simple Object Access Protocol), which is a Web services execution protocol, according to the service plan. This paper is organized as follows: 1) Brief introduction to the Semantic Web Services technologies as fundamental background for the paper; 2) Description of the detailed design of ubiHome architecture; 3) Explanation of the prototype implementation and experiments in our home network test bed; 4) Discussion of some limitations and future works as conclusions.

Young-Guk Ha received his PhD degree in computer science from Korea Advanced Institute of Science and Technology(KAIST). He joined Electronics and Telecommunications Research Institute(ETRI) in 1995 and is currently a senior member of engineering staff in Intelligent Robot Research Division. His research interests are in ubiquitous computing and service robotics.

 Isochronous resource reservation for the deterministic delay guaranteed collaboration 

Seong-Soon Joo (ETRI, Korea); Sangjoon Park (ETRI, Korea); Cheol Sig Pyo (ETRI, Korea); Jong-Suk Chae (ETRI, Korea)

With increasing processor speeds and convergence on wired-wireless networks, the potential for computing complex behavior in real-time collaboration has greatly improved. In recent years, distributed virtual environments have become a major trend and application fields of collaboration have been diffused into machine-to-machine services. Collaboration requires communication channels among resources and coordination functions for the integration and harmonious adjustment of the individual work effort towards the accomplishment of a goal. For realizing the advanced collaborative environments, highly interactive communication will be indispensable. New features from Ethernet technology to meet delay and jitter requirements for time-constrained collaboration will be required. In this paper, to obtain the deterministic latency transport for supporting the time-constrained collaboration, Ethernet frame forwarding with the isochronous resource reservation is proposed. We define the isochronous forwarding window which is the time duration and virtually identified on point-to-point links, and propose the resource reservation in terms of isochronous forwarding window with the admission algorithm based on probabilistic admitted level of quality for the collaboration communication channel.

Seong-Soon Joo received his B.S. from the Hanyang University in 1980, and received M.S. and Ph.D. degree from the Seoul National University, Korea, in 1982 and 1989 respectively, all in electrical engineering. He joined ETRI (Electronics and Telecommunications Research Institute) in 1983, where he is the principal investigator in Ubiquitous Sensor Network Division. From 1996 to 1997, he was a visiting research faculty at Computer Science and Engineering Dept. of Arizona State University, USA. Since September, 2004, he is an adjunct professor at the Dept. of Broadband Network Engineering, University of Science and Technology, Taejon, Korea. He has worked in a range of fields, including the development of packet switching and frame relaying for ISDN switching system, call control software for ATM networks, architecture design of high-speed IP router, and the development of all-optical
cross-connect system focused on lightpath provisioning, protection, and restoration. His research interests include soft computing, active network, intelligent control for communication networks, and network architecture for the post-Internet era.

**Small, Embeddable Web Server for Home Appliances with MPU and Real-time Operation System**

Masato Shimano (Ryukoku University, Japan); Futoshi Okazaki (Renesas Solutions Corp., Japan); Yoshihiro Saito (Renesas Solutions Corp., Japan); Akiya Fukui (Renesas Solutions Corp., Japan); Takako Nonaka (Ryukoku University, Japan); Tomohiro Hase (Ryukoku University, Japan)

This paper describes a small, power-saving, and embeddable web server for home appliances. The proposed device has an embedded MPU and a real-time operating system to actualize small size and power-saving for consumer use. First, a T-Engine, embedded 32-bit RISC MPU was used as the hardware of the proposed system. Next, a real-time operating system designed for embedded use, T-Kernel / Standard Extension, was selected. Thirdly, a TCP/IP protocol stack and applications were developed. The prototype has 1/40 or less of the electrical power consumption of a general PC server and the software resources are approximately 1.6 MB. Finally, in order to verify the prototype system, performance tests using a general web service and a remote manipulation function via the Internet were conducted. As a result, the display of text and dynamic picture images, and remote manipulation of the server from another terminal inside the network were verified with poise.

Masato Shimano was born in 1982 in Osaka, Japan. He received a B.S. degree from Ryukoku University, Japan, in Media Informatics in 2007. He has been a student of the graduate school of Ryukoku University since 2007. His major is media informatics and communication.

**A Service-based QoS Provisioning Framework for the Home Networks**

This paper presents an efficient service-based QoS framework (SQF) to provide end-to-end QoS provisioning services. The SQF provides service bandwidth reservation services as well as priority-based QoS application services on Ethernet networks. The SQF works on a home gateway and legacy networking devices. Although the SQF can support priority-based QoS services, in order to provide high-quality multimedia services to home residences, our scheme supports the service bandwidth reservation mechanisms because different home applications from multi-rooms compete for limited bandwidth resources. When application session setup begins, our SQF negotiates a QoS service level of applications through push/pull based QoS APIs. Application services can perform any number of dynamic QoS negotiations with SQF before the service starts. Thus, the QNSF can provide well-adapted QoS providing service to home multimedia applications. In addition, our framework can cooperate with the UPnP QoS architecture by parameter mapping. The SQF consists of some functional blocks. A QoS client module defines push/pull based QoS negotiation APIs for home service applications that require QoS negotiations. QoS proxy, QoS manager, and traffic controller modules support service-based QoS provision functions like dynamic bandwidth negotiation and priority assignments, QoS admission control, service traffic rate control, and adaptive QoS parameter mapping. Packet classifier and class marker modules enable dynamic input traffic classification and support policy based dynamic rate controls. Our implementation and its performance evaluations show the feasibility of the SQF for the provisioning QoS-based application services in bi-directional multi-room home networks. In addition, our SQF supports flexibility for the interoperability with the UPnP QoS architecture.

**Video-on-Demand Streaming in P2P Environment**

Jong-Hyuk Roh (ETRI, Korea); Seung-Hun Jin (ETRI, Korea)

Recently, there have been several research projects on live streaming using P2P approach. However, applying these techniques into VoD streaming is not a trivial task due to the following fundamental differences between the two types of streaming. First, end-to-end delay is more important to live streaming than VoD streaming. In live streaming, the shorter the end-to-end delay is, the more lively the stream is perceived by the users. In VoD streaming, liveness is simply irrelevant because the video stream is already pre-recorded. This fact implies that while a short tree rooted at the video server and spanned over clients is desirable in live streaming, it is not a necessary condition for the case of VoD streaming. Second, a user joining an on-going live streaming session is only interested in the stream starting from his/her joining time, while in the VoD streaming case the whole video must be delivered to the new user. As such, a good VoD system must find an efficient way to provide the initial missing part of the video to the latecomers. Moreover, the correlations between various variables are different for the two types of streaming. For example, a user will likely stop watching a VoD stream when its QoS degrades, but the user may not do the same thing for a live stream because he/she doesn't have an option of watching it again in the future. Therefore, it is expected that if the QoS of the video stream reduces, many more clients will leave the system in the case of VoD streaming than in the case of live streaming. This observation stretches the importance of a quick and localized failure recovery protocol in a VoD streaming system. It is critical for one to take those differences into account in order to build a successful VoD streaming system in a P2P environment, if the proposed technique is a variant of any existing live streaming systems. Our proposed technique in this paper is not
based on any existing live streaming systems; however, the aforementioned differences are still useful for us to realize challenges in building a P2P VoD streaming system. We consider the following problems as dominant to be solved for a P2P VoD streaming system: (1) quick join; (2) fast and localized failure recovery without jitter; (3) effective handling of resources’ attributes; and (4) small control overhead.

**Home Entertainment 2**

Room: Trinity 4

**Game System using Facial Features for the Handicapped people.**

Ju Jin sun (Konkuk University, Seoul South Korea, Korea); Eun Yi Kim (Konkuk University, Korea); Yunhee Shin (Konkuk University, Seoul South Korea, Korea)

Recently, computer games using traditional interface such as a keyboard and a mouse have been replaced by new game paradigm such as body-interaction games. The body interaction games use human’s gesture to control a game, which makes the players feel more realistic enjoyment and actual feelings. From now on, various interfaces based on human gestures have been developed to provide the natural communication in between players and game systems. Although such systems are well worked on the people without physical disabilities, they can not be applied to the handicapped people. For the handicapped people, the game system with general interface that adaptable to the people with various physical disabilities should be developed. In this paper, we develop an interactive game system that controls a game using only the movement of human’s facial features. Our system is specially designated for the handicapped people with severe disabilities and the people without experience of using the computer. Using a usual PC camera, the proposed game system detects the user’s eye movement and mouse movement, and then interprets the communication intent to play a game. The game is made using FLASH (version MX 2004), and is one of the shooting games where a user hits randomly flying bird with a gun. A player controls the cursor by his (or her) eye movement and fires a gun by opening his mouth. These movements of a player are detected and tracked in the interface modular. firstly extracts user’s face from the background using skin-color model, and then localizes user’s eyes and mouth from the face region. To be robust to the complex background and users with various physical conditions, eye regions are localized using neural network (NN)-based texture classifier that discriminates the facial region into eye class and non-eye class, and then the mouth region is detected based on edge information. Once these features are extracted, they are continuously tracked by facial feature tracker: a mean-shift algorithm is used for eye tracking, and a template matching is used for the mouth tracking. Based on the tracking results, mouse operations such as movement or click are implemented in a mouse controller. Our game system moves the cursor to the point at which the user gazed on the display, and then fires the gun to that point if he opens and closes his(or her) mouth. To show the effectiveness of the game system, it was applied to the disabled users. The time to play a game could be rapidly reduced if the disabled users would have the sufficient practice. The experimental results show that our game system should be efficiently and effectively used as the interface for the disabled people.

**Trained Hybrid Filters for Image Upscaling**

Ling Shao (Philips Research Laboratories, The Netherlands)

Currently, we witness a major revolution in video display technology. Flat screens based on, either Liquid Crystal (LCD), or Plasma (PDP) technology in most consumer applications are rapidly replacing the Cathode Ray Tube (CRT) that served as the workhorse for almost a century. For television, an important consequence is that the old trade-off between light-output and resolution has been eliminated. This breakthrough facilitates the transition from standard definition television (SDTV) to high-definition (HDTV), but we expect SDTV and HDTV formats to co-exist for quite some time. Therefore, the question arises how to optimally display legacy video on modern screens. Linear up-scaling techniques, such as nearest neighbor, bi-linear, and bi-cubic [1] interpolations, have been popular in many applications. However, these linear methods usually result in blurred images, because the scaling process does not add new frequency components. Recently, several advanced non-linear resolution up-conversion algorithms have been developed. Li and Orchard [2] proposed an edge-directed interpolation method, which assumes that the orientation of a local edge does not change with scaling. The filter coefficients are approximated from the low-resolution image within a local window by minimizing the Mean Square Error (MSE). Due to the fact that the coefficients optimization is done on the fly, this method is computationally very demanding. A classification-based interpolation algorithm was proposed by Atkins et al. [3]. The classification is carried out in the feature space based on the Expectation Maximization (EM) method. A training process, using Maximum Likelihood (ML) estimation, is done to get the optimal filter coefficients. The above mentioned advanced resolution up-conversion algorithms are either computationally too expensive or not so satisfying in performance. In this paper, a hybrid filter is proposed for image resolution upscaling. The hybrid filter is composed of a linear filter and an order statistic filter. In the linear filter, the pixels are ordered spatially, and in the order statistic filter, pixels are ordered according to their pixel value differences to the central pixel. The local content of the image is classified by...
the combination of structure and activity measure. The structure information is represented by Adaptive Dynamic Range Coding, which is a simple and efficient coding method for structure. Several activity measures, such as variance, entropy and dynamic range, are employed. The optimized filter coefficients for each class are obtained by an off-line training process, which trains on the combination of the original high-resolution images and the down-sampled versions of the original images. During the filtering process, the optimized coefficients are retrieved from the look-up table for image interpolation. For evaluation, the proposed hybrid filter is compared with several state-of-the-art image interpolation algorithms both subjectively and objectively. The experimental results show the superior performance of our proposed technique. For future work, the proposed method could also be applied on other video processing scenarios, such as sharpness enhancement, noise reduction, and compression artifacts removal. [1] E. Meijering, “A note on cubic convolution interpolation”, IEEE Trans. Image Process., Vol. 12, No. 4, pp. 477-479, Apr. 2003. [2] X. Li and M. T. Orchard, “New edge-directed interpolation”. IEEE Trans. Image Process., Vol. 10, No 10, Oct. 2001, pp. 1521-1527. [3] C. B. Atkins et al., “Optimal image scaling using pixel classification”, Proc. ICIP 2001, Vol. 3, pp. 864-867, 2001.

**Design and Implementation of Low-price Name Card-Size Portable Digital TV through High-Integration**

Y. Bai (Fu Jen Catholic University, Taiwan); Ming-Huei Lin (Fu Jen Catholic University, Taiwan)

A survey of the digital TV products on the market shows that at present there are still many disadvantages to be encountered, such as too much power consumption, high temperature and poor signal quality. Hence we propose a new design with low power consumption, low temperature and better signal quality. To reduce power consumption we select a high-efficiency DC/DC converter as the power source for the whole system. The chipset selection includes an LCD driver for the LCD panel backlight circuit. In addition we select a high-efficiency class D power amplifier to reduce power consumption and to simplify the peripheral circuit. Moreover, we compare the power consumption of the various high-integration DVB decoder chipsets to further reduce the power consumption.

At the same time, by selecting appropriate solution and key components, we minimize both size and cost of the hardware circuit. Facing the cumulative heat problems, we propose a very thin “heat sink” which can conduct heat to the surrounding air. By means of IR scanning we have found that the “heat sink” can reduce the temperature of the outer case from 48°C to 39°C while the TV set is running for 24 hours. Antenna efficiency and DVB-T RF receiver performance act as two key factors of the signal quality. To avoid signal reflection, the RF trace on the PCB is thoroughly impedance controlled at 75 Ohm. To improve the sensitivity and the dynamic range of the receiver, we place a low-noise amplifier (LNA) with low noise figure (NF), better linearity and wider bandwidth in the front-end of the RF part and connect a low-pass filter (< 0.5 dB loss) as a pre-selector between antenna and LNA. The low-pass filter decreases the interference from out-band noise. We connect the pi-type filter to the power source to reduce power ripple and protect the DVB-T RF receiver from radiated emission by covering it with a shield case. By using 64-QAM modulation our DVB-T RF receiver sensitivity is -80 dBm, RF dynamic range is 60 dB and NF is 5-6 dB. As for the antenna module, because the channel bandwidth of digital TV is wide, our antenna design must have broadband characteristics. Hence we select a planar printed antenna, which has the advantages of minimizing the antenna size, keeping enough power gain in the broadband, centralizing the input impedance and maintaining near circular polarization of the omni-directional E-field pattern. Overall, our design currently has numerous competitive advantages: low power consumption at 2.6Watt, low temperature at 39°C, better signal quality with a built-in antenna support for DVB-T VHF and UHF band, battery replacement, high mobility (90-100 km/H), low cost (USD 58), only 120g weight (with battery) and small size, the dimensions being 101mm(L) x 56mm(W) x 20mm (D).

Ying-Wen Bai is a professor in the Department of Electronic Engineering at Fu-Jen Catholic University, Taiwan. His research focuses on mobile computing and microcomputer system design. Ying-Wen Bai obtained his M.S. and Ph.D. degrees in electrical engineering from Columbia University, New York, in 1991 and 1993, respectively. Between 1993 and 1995, he worked at the Institute for Information Industry, Taiwan.

**Design of the Autonomous Fault Processing Mechanism for Home Network**

Young-Sung Son (ETRI, Korea); Tai Yeon Ku (ETRI, Korea); June Hee Park (ETRI, Korea); Kyeong-Deok Moon (Electronics and Telecommunications Research Institute, Korea)

Abstract This paper proposes the design of the autonomous fault processing mechanism that can be used to solve abnormal faults in home network. In home network environments that consists of several kind of networks and devices, their unrevealed functionalities can cause non-instinct faults (combination or inferred situation). Usually many researches had tried to make a model-based fault processing mechanism. But, those works depended on the characteristics of the model for the specific environment. We focus on the process in which faults are induced and establish fault categories that can be caused in home network and the autonomous fault processing mechanism. Problem The integration of computing-power and communication into the next generation of everyday device is one of the major factors that is driving the emergence of home network, which interconnects various appliances, enabling remote access to and control of those appliances, and any available services such as home

http://edas.info/showProgram.php?_qf__showProgram=&c=5136&form...bio%5D=1&program_view=everybody&action=Show+conference+program
entertainment, home office, and home automation. Nowadays, we can find the increasing diversity of home devices and various kinds of home network middleware that are working in home. Typically, home network environment has a special feature that there is no administrator. If faults occur, user can not response the suitable fault processing and can induce another faults.[1] Our solution We need to develop fault detect, diagnosis and recovery system in home network. First of all, we classify the generic fault categories shown in table.1 and design the home network fault processing mechanism described in fig.1. Our approach is (1) introduction of a fault model as specification of the home network environment, (2) integration of the fault model and fault processing mechanism, and (3) development of an stand-alone fault management software in home gateway.

**Power Monitoring and Control for Home Appliances Based on Power Line Communication**

Chia-Hung Lien (National Taiwan University of Science and Technolog, Taiwan); Hsien-Chung Chen (Fu Jen Catholic University, Taiwan); Y. Bai (Fu Jen Catholic University, Taiwan); Ming-Bo Lin (Department of Electronic Engineering, National Taiwan University of Science and Technology, Taiwan)

Power monitoring and control through home networks is becoming important for home appliances. Radio, infrared, metallic cable and power line are popular medium for home networks. They are used according to capability of transmitting range, ease of installation, communication reliability and cost. Radio communication holds wireless and mobility as features. Wireless LAN (802.11b) and Bluetooth attract attention in respect to high speed and low cost. Infrared communication is widely used for remote control for home appliances, such as television and air-conditioner. Few regulations, no interference with the neighborhood and high-speed communication capability are featured. Metallic cable communication is used for a system where transmission stability qualities are required. Power line communication (PLC) is used to utilize a domestic power line as a communication cable. Because PLC is seen as a no-new-wire home network medium, it is easily installed into an existing existence. A no-new-wire embedded system to monitor/control power for home appliances remotely has been developed in this paper for home power management. By using PLC technologies, home appliances can be controlled and monitored through AC wire lines. The no-new-wire embedded system consists of three blocks: PPCOM (PLC Power-Controlled Outlet Module), embedded home server and remote control. The PPCOM combines with the multiple AC power sockets. In the PPCOM, a simple plug-in mini-controller performs the power on/off switching of the sockets. The PPCOM also includes a power detection circuit to verify the power on/off status of home appliances and to measure power consumption with power factor. The embedded home server is built with an embedded platform which is designed and controlled by software using Visual C++ Studio 2005. That supports the Graphic User Interface (GUI) to let the user easily monitor and control the power on/off of the home appliances. We penetrate the PLC to receive/transmit between the embedded home server and the mini-controller in the PPCOM. The embedded home server offers an automatic system for power monitoring and control. It also collects information for analysis of power consumption including daily, weekly, monthly electricity consumption of individual appliance and total consumption of Home. Furthermore, because the low power factor loads increase losses in use of power and results in increased cost for electrical energy use, the detection circuit also measures the power factor of individual appliance. The remote control mechanism allows the user to monitor and to control the power on/off of home appliances through the Internet. For introducing the PPCOM to a home network, engineering time for installation is about 0.5 hour conventionally. Our system is developed and installed in our Lab to control air-conditioners, lights and other appliances for evaluation. The embedded home server actively enforces power off for the appliances in action and notifies the user to turn off the idle ones, achieving a power conservation of about 23%.

**Mobile Computing & Communication 3**

*Room: Trinity 6-8*

**Interactive DMB System – Field Trial**

Markus Mehnert (Technische Universität Ilmenau, Germany); Eckhardt Schön (Technische Universität Ilmenau, Germany)

Digital multimedia broadcasting (DMB) is a service that is based on digital audio broadcasting (DAB). The Institute for Media Technology operated a DAB test transmitter for evaluating simple DMB applications in the last two years. Now we start a field trial at the campus of our university aiming at interactive applications. The feedback channels will be the university intern local area network (LAN) and wireless local area network (WLAN). In this paper we present the technical requirements needed for operating the whole system. One part will be a prototype USB (universal serial bus) receiver that provides the data stream to the connected personal computer. Furthermore tested applications and new applications will be discussed.

**An Extended T-DMB BWS for User-friendly Mobile Data Service**
Hee-Jeong Kim (ETRI, Korea); So Ra Park (Electronics and Telecommunication Research Institute, Korea); Bong Ho Lee (ETRI, Korea)

In this paper, we have proposed DMBWeb, an extended T-DMB BWS, which supports a multi-modal user interface for easy and friendly access of a mobile user. To realize it, DMBWeb have defined a new contents format, VeBWS, and developed a dedicated authoring tool for the VeBWS content. Because the VeBWS content specification simply adds voice interface information to existing BWS html specification, there is no surprise increase in the size of the data or processing overhead required. DMBWeb users conveniently access data service by multi-modal user interface. The main characteristics of the authoring tool are to separate contents into data and presentation for reusability, to provide a form to input data for productivity, and to generate content documents automatically while combining the data document and presentation template. Furthermore, content providers can expect the additional gains by selling presentation template. Finally, we show that the proposed protocol operates stably and effectively in the terminal with MOT decoder and VeBWS browser, through the experimental broadcasting in the local DMB network.

Heejeong-Kim received the M.E degree in Computer Engineering from ICU, Korea in 2000. She is a senior researcher at Electronics and Telecommunications Research Institute, Daejeon, Korea. She has worked on a data broadcasting service, including BWS and middleware platform. Her research interests are a mobile broadcasting systems and a conditional access system.

Data Broadcasting Server and Receiver for Interactive Data Services based on Middleware in Terrestrial DMB

GwangSoon Lee (ETRI, Korea)

In this paper we present a development of the data broadcasting server and receiver for interactive data services based on middleware technology. Terrestrial Digital multimedia broadcasting (T-DMB), T-DMB provides a mobile multimedia broadcasting services in good quality, based on the Eureka-147 DAB system. In addition to the video standards including MPEG-4 technologies, T-DMB supports the various data transport protocol such as Multimedia Object Transfer (MOT), IP Tunneling, and Transparent Data Channel (TDC), which is specified by Eureka-147 DAB. Recently, T-DMB also accommodates the interactive data services based on T-DMB middleware that was newly designed by using the JAVA technology. The middleware platform for data broadcasting in T-DMB, called T-DMB MATE (Mobile Application Terminal Environment), defines a context for running the applications and APIs so that the various applications can provide the high-level functions for the interactive data service. Specially, T-DMB MATE is designed to be suitable for the channel environment of mobile broadcasting, supporting scheduled data broadcasting/telecommunication network conditions. A variety of T-DMB receiver will be available for this service and then transmits real time data with a few channel capacities. The system structure of broadcasting system for data services based on T-DMB MATE consists of an authoring tool producing T-DMB MATE contents, a data agent processing live data needed at the contents, a broadcasting data server for creating the signaling messages and for transmitting all of them. The T-DMB MATE provides bidirectional data services to users using the mechanism in which Java-based applications are downloaded through broadcasting or telecommunication network and a small amount of live data additionally needed for their behavior is transmitted at real time. Therefore, Java based applications can be created by authoring tool and live data such as stock, weather, news related to these applications can be collected by data agent. The data broadcasting server gives database functionality managing T-DMB middleware contents and effective scheduling mechanism taking account of service scenario or broadcasting/telecommunication network conditions. A variety of T-DMB receiver will be available for this service by porting the MATE technology into them. This paper begins with the overview of interactive data services in T-DMB MATE, followed by its implementation including the data broadcasting server and T-DMB receiver that can be used for the T-DMB MATE data services. The major functions of the implemented data broadcasting server is to manage applications with module unit-common modules, specific modules, or real-time datum (news, stocks, etc), which is the key characteristic of the T-DMB MATE. Moreover, the data broadcasting server is able to set up a transmission plan in order effectively to use broadcasting bandwidth, transporting the signaling message and application modules according to the schedule. The developed data broadcasting server is comprised of the Contents Management Server, the Transmission Server, and the Database. The Contents Management Server performs the scheduling function to transmit the application signaling messages or application modules during the constant time interval of designated date. Using this scheduling information set up at the Contents Management Server, the Transmission Server transfers the application modules and signaling messages, and performs the function of monitoring the sending state. The structure of the implemented T-DMB MATE receiver is made up of the RF tuner, Baseband Processor and Media Processor that includes the video and audio decoder. All the functions of T-DMB MATE are performed in the Main Processor having WinCE as its operating system. The T-DMB MATE is implemented with four layers of the DMB MATE API implementation layer, JVM interface layer, middleware engine layer, and porting layer. The DMB MATE API is designed by Java-language. However, most of the low-level running engine is implemented by C-language rather than Java-language for several reasons, such as direct control of the hardware device and rapid computation. We have also made various types of applications...
by using the standardized MATE APIs. By using such applications, we verified the developed data broadcasting server and receiver under various conditions including the experimental broadcasting service with a commercial broadcaster. Therefore, it is certain that T-DMB MATE will provide high-level interactive data service in mobile environment in which all the new technologies are being converged.

**A RF/Baseband/Multimedia Single Chip Implementation of T-DMB Receiver**

Bontae Koo (ETRI (electronics and telecommunications research institute), Korea); Juehyun Lee (ETRI, Korea); Jin kyue Kim (ETRI, Korea)

The mobile video, audio and data services can be provided by a T-DMB (Terrestrial Digital Multimedia Broadcasting) system based on Eureka-147 standard in Korea. In this paper we show the design and implementation of an integrated T-DMB Receiver SoC which includes RF tuner, Baseband decoder and Multimedia decoder. T-DMB uses the VHF channel from 7 to 13 of 174 MHz to 216 MHz which are in Band III. A 6 MHz bandwidth of the channel is divided by 3 ensembles. Each ensemble has a 1.536 MHz bandwidth and can service the audio, mobile TV or data. A T-DMB receiver is largely consisted of a RF tuner, Baseband decoder and Multimedia decoder. The chip size is 8.5m x 8.5m in 0.18um CMOS RF technology, and it has a low power consumption. This RF tuner is integrated by using RF-CMOS technology. RF input frequency range is from 170 MHz to 240 MHz and IF output frequency is 38.912 MHz or 2.048 MHz. This RF tuner includes a LNA (Low Noise Amplifier), RF AGC (Automatic Gain Control), RF down-conversion mixer, a VCO (Voltage Controlled Oscillator), a frequency synthesizer. A fully integrated frequency synthesizer is a sweeper with the typical channel step of 16 KHz which can cover European DAB (Band III) and Korean T-DMB band. I2C serial interface is used for frequency synthesizer and other function setup. The RF input frequency range is 170~240 MHz, and power consumption is approximately 70mW. The baseband decoder includes QDD, AFC, AGC, FFT and differential QPSK demodulation and frequency and timing de-interleaving and Viterbi decoding and inverse energy dispersal de-scrambling and Reed-Solomon decoding and convolutional de-interleaving for error correction for mobile multimedia data, and dedicated HW MP2 audio. The functional blocks are divided into largely 3 parts system demuxing, video decoding and audio decoding. System demuxing part consists of TS interface and TS manager, RISC processor, and parser logic which can connect to SDRAM controller. System demuxing part receives transport stream and demultiplex T-DMB stream and store the parsed elementary stream into SDRAM. We implemented single chip T-DMB Receiver SoC. The developed Single chip T-DMB SoC performed 30 frames/s of CIF at 27MHz in real-time. The chip size is 8.5m x 8.5m in 0.18um RF CMOS technology. The proposed architecture brings about high cost-effectiveness and low power consumption for the portable T-DMB. *The specification of T-DMB SoC Parameter RF Baseband Multimedia Technology 0.18um CMOS 1P6M Chip Size 8.5 x 8.5 mm2 Main Clock 24.576M 24.576M 54M Power Supply 1.8V 3.3/1.8V 3.3/1.8V Power Dissipation ~70mW ~90mW ~90mW*

**Development of a CDMA Converged T-DMB Receiver for Interactive Broadcast Web Site Services**

Byungjun Bae (Electronic and Telecommunications Research Institute (ETRI), Korea); Joungeil Yun (Electronics and Telecommunications Research Institute (ETRI), Korea); Yong-Tae Lee (Electronics and Telecommunications Research Institute, Korea); Jong Soo Lim (Electronics and Telecommunications Research Institute, Korea)

T-DMB (Terrestrial Digital Multimedia Broadcasting) is one of the applications that have emerged from the European Eureka-147 DAB (Digital Audio Broadcasting) standard. Particularly in Korea, T-DMB has been developed and commercialized for multimedia broadcasting services in various mobile and portable environments. Therefore, in addition to both audio services of Eureka-147 DAB (Digital Audio Broadcasting) standard in Korea, T-DMB had been developed and commercialized for multimedia broadcasting services in various mobile and portable environments. In this paper, the protocols for transmitting data contents in T-DMB are explained and a new scheme for interactive T-DMB data services with telecommunications network is proposed. Especially, the proposed interactive data service scheme is used to adapt the BWS (Broadcasting Web Site) service, which is originally a unidirectional application, for interaction with contents from communication network. The traditional BWS service is to provide a similar service as the internet web service through the T-DMB network, which uses the transfer protocol named MOT (Multimedia Object Transfer). Since the T-DMB, a sort of a broadcasting system, has the limited data transfer rate, it is actually impossible to accommodate all web data that users want. Therefore, this paper proposes the interactive BWS service that provides various data, which users request after receiving the data from T-DMB network, through the telecommunication network. The data received from T-DMB and telecommunication networks are linked by the address of URL in contents. To verify technically the proposed service scheme, we designed and implemented the interactive T-DMB receiver with the CDMA module as hardware and software. This receiver has two API set: one is for operating T-DMB services and the other is for linking the CDMA network. We also implemented and operated the data broadcasting server and the return...
channel server sending data that contain various data contents. The experimental result shows the possibilities of new interactive services in the digital broadcasting field by utilizing the mobility of T-DMB and CDMA.

12:30 PM - 1:30 PM

Keynote 6★

The Intelligent Home
Room: Trinity 5

1:30 PM - 3:00 PM

Consumer Networks 6★

Room: Trinity 1-3

A Novel Adaptive Unequal Error Protection Method for Scalable Video over Wireless Networks ★
Amir Naghdinezhad (University of Tehran, Iran); Mahmoud Reza Hashemi (University of Tehran, Iran); Omid Fatemi (University of Tehran, Iran)
With the rapid development of wireless networks, advanced multimedia services such as video phone, video conferencing and video streaming are becoming mainstream. In these applications, devices such as phones and laptops transmit or receive compressed video signals over wireless or mobile networks. Although video communications are particularly attractive in wireless systems, there are subject to the limitations of these networks. Limited channel bandwidth and high sensitivity to noise are the two major challenges of wireless channels. To address the limited bandwidth we need a better compression performance. Bandwidth is especially a concern when multiple clients with different specifications in terms of available bandwidth, allowed power consumption and display resolution simultaneously access the same compressed content. Noise sensitivity may cause both packet loss and bit errors which result in a reduced quality of service (QoS). Scalable video coding (SVC) methods like the scalable extension of H.264/AVC can be employed for solving these challenges in wireless networks. By using scalable video coding, an efficient flexible bit stream is produced which alleviates the effect of network heterogeneity. This single bit stream contains the information to fulfill the requirements of different clients. Furthermore, since the video layers in an SVC stream have different importance, unequal error protection (UEP) is applied on the video signal. Applying UEP on scalable video signal improves the efficiency and reliability of the network. In this paper, we propose a protection method for enhancing the quality of scalable video over wireless networks for a wide range of error rates. The proposed method makes use of Forward Error Correction (FEC) schemes based on Reed Solomon codes for unequal error protection. The scalable extension of H.264/AVC is chosen as the encoder module because of its higher efficiency in comparison to other scalable standards. First, the amount of required extra data for each temporal-SNR layer is calculated. This calculation is based on SNR level, temporal level, rate and the quality improvement caused by correctly receiving the corresponding part. Next, the extra data is added and the packetization scheme is performed. This process is performed for each group of pictures (GoP) independently. The proposed method has been tested against various error, and packet loss scenarios. Standard H.264/AVC sequences in both CIF and QCIF format were used. Experimental results show a significant improvement of 5dB in average, in comparison with conventional equal error protection (EEP) methods. The proposed method is far less complex compared to other existing methods like genetic algorithm (GA) and hill-climbing which use an UEP approach. The lower computational complexity of the proposed method makes it suitable for mobile devices where power consumption, and processing power are always a concern.
Mahmoud R. Hashemi received his B.Sc, and M.Sc. in Electrical Engineering from University of Tehran. He pursued his Ph.D. at the University of Ottawa, Canada. He is currently an assistant professor, and the director of the Multimedia Processing Laboratory at the University of Tehran. His research interests include very low bit rate image and video coding, compressed domain processing, and hardware implementation of image and video coding standards.

SCE Library Implementation for Parlay X APIs working on both MS .NET & Java open platform ★
Seung-Hwa Chung (ETRI, Korea)
Abstract - To create the new open service, service developers had to know very low-level techniques of telecommunication network and it tool long time period, but now it's not. Because Open API separates

http://edas.info/showProgram.php?_qf__showProgram=&c=5136&form...bio%5D=1&program_view=everybody&action=Show+conference+program
telecommunication network into two parts; service layer and network control/transport layer. Open API concept made the creation of the open service independent from the low-level telecommunication structure. Parlay has appeared followed by Open API idea. Open API standardization was initiated by Parlay group in 1998, and the work of Parlay group was followed by 3GPP. 3GPP made new working group called OSA (Open Service Architecture/Access) to define Parlay APIs. Later, Parlay group, 3GPP, and ETSI made joint-API-group to define common API specifications. Parlay APIs has become the standard open service interface of open telecommunication network, so the existing service providers and 3rd party service providers can access and share resources of the telecommunication network by the standard open service interface. This also brings the solution of expanding business model of the telecommunication network provider. To satisfy customers’ desire and to cope with the convergence of today’s network, open service creation model need to be fast and have short life time. The developer is required to create open service fast and many, and this invokes the better SCE (Service Creation Environment). Parlay X has born to simplify Parlay, and further SCE are developed to use parlay X APIs even more easily. Parlay is working on CORBA tech., and Parlay X is on Web Services tech. Today, Web Services is very well-known as a middleware that can inter-communicate between many different program languages. This paper, we write the result of the implementation of SCE Library for Parlay X APIs. This SCE Library is tested and verified on Parlay X Gateway Simulator, developed by BcN Service Research Group, ETRI. Parlay X Gateway can be made by one of various platforms, and we implemented SCE Library works for MS .NET and Apache AXIS platform based gateways. Because above two platforms are most used for Web Services tech.

As this is paper abstract, we will briefly write about the implementation process. 1. Get published standard WSDL about Parlay X APIs. 2. Create proxies and stubs for Parlay X APIs. 3. Add proxies and stubs for the Utility APIs that are useful for the internal program control and Internet related utilities. 4. Implement SCE Library in separate program language C# and Java to give the developer the option to select the program language depending on the developer's system environment. After Implementation of SCE Library is completed, we examined it. […] SCE Library successfully worked on both MS .NET platform based gateway and Apache AXIS platform based gateway. We confirmed that Parlay X APIs can successfully inter-communicate between MS .NET and Apache AXIS platforms by the implementation of SCE Library for Parlay X APIs.

Performance of Triple Play Services in Wireless Meshed Networks
Mikael Gidlund (Royal Institute of Technology, Sweden)

Wireless mesh networks (WMNs) will play an important role in the next-generation wireless communication systems because it can support broadband services with ubiquitous coverage by low transmission power. In this paper, we will discuss and evaluate triple play (VoIP, Internet and IPTV) services in terms of packet loss and delay over a wireless meshed network.

Next Play Evolution: Beyond Triple Play and Quad Play
John Ulm (Fellow of the Technical Staff, Motorola Connected Home Technology Office, USA); Bill Weeks (Motorola, Inc., USA)

Service providers have been very successful in offering Triple Play services to its customers and look to equal success in offering Quad Play services in the near future. Looking under the hood, this has primarily been a bundling process with the underlying applications of video, voice and data remaining relatively isolated in their own silos. This world is about to radically change. On the horizon is a host of mobile multimedia devices that will be integrated into today’s home video experience. This will cause a slew of new applications that will blur the boundaries between voice, video and data. This paper discusses a vision of Converged Services where users can access any content, from any device, whenever and wherever they want. This will generate many new services (and new revenue streams), which will make the concept of Triple Play and Quad Play no longer relevant. We need to start thinking in terms of the Next Play and the ability to quickly add many new services. Converged networks within the home will become a key element in tearing down the silos between the video/voice/data worlds. Devices will be interconnected with high speed, broadband capable home networks, such as MoCA®, HPNA and PowerLine, to enable the “any content to any device” mantra. Connecting the video world with the data world enables new sources of video content from the IP world (sometimes referred to as IP video, IPTV or Long Tail content). The converged network will also bridge between the wired & wireless worlds with technologies such as 802.11 b/g/n and Bluetooth. This wireless bridge to mobile devices can then be extended from inside the home to reach throughout the community. The converged network enables a distributed approach of resources that can be used to provide a seamless experience, a single virtual media environment in the home. IMS is often mentioned as the framework for providing converged services. A standards based infrastructure that IMS/SIP offers enables a rapid development of new applications. With a standard IMS control infrastructure and a converged home network in place, it then allows existing applications migrate to this new framework to further
enable any content to any device. IMS/SIP will be the glue that binds across them all to enable the Next Play experience.

**An Integrated Service Management Scheme for the Digital Home Environment**

Jong-Hoon Lee (ETRI, Korea)

This paper proposes a mechanism for service management in the digital home. It helps for users to install, download and execute services based on Open Home network Framework (OHF) which provides a platform technology to manage user applications, home network middleware, and interoperability among consumer electronics (CE) in the open home digital home environment. To provide interoperability and coordination among various home services, we provide a service adapter within the OHF. Although the conventional home gateway system was considered necessary device for managing various devices and services at home network, we have implemented the service adapter in the home server [1]. The home server provides broadcasting services including IPTV, digital HDTV, CATV, home video conference and multimedia services such as media sharing service as well as home-automation and home appliance control services of the traditional functions of the home gateway system. Generally, the existing home server limits users to download and execute intended service applications when additional services are available from the service provider. It causes users can only use local services which were previously installed on the system [2]. Thus, an additional scheme is required on the home server to execute the service packages after the users download intended service packages from service providers in the external network. The proposed service packaging mechanism enables not only to install and operate available service packages on the home server, but also users to use various services from external service providers. Also, most of digital home services are consist of an upper service UI for users and a lower legacy service that are invoked upon requests from the service UI. In order to achieve interoperability among various services, we propose a Service Adapter within the OHF. By doing so, the service interoperability between the upper service UI and the lower legacy service can be achieved.

**Mobile Computing & Communication 4**

Room: Trinity 6-7    Chair: Mehrdad Nourani (University of Texas at Dallas, USA)

**Optimizing the DVB-H Time Interleaving Scheme on the Link Layer for High Quality Mobile Broadcasting Reception**

Michael Kornfeld (Braunschweig Technical University, Germany)

The DVB-H technology (Digital Video Broadcasting – Handheld) extends mobile connectivity of handheld terminals, specifically offering mobile television and other broadcasting services to small portable devices. This paper presents a performance analysis and a proposal for a parameter optimization of the time interleaving scheme included in the DVB-H forward error correction. Various sets of parameters are studied and optimized with respect to the link performance. An enhanced time interleaving scheme is proposed and evaluated. The channel characteristics used for the analysis correspond to typical mobile usage scenarios which are challenging for the forward error correction.

Michael Kornfeld received his Dipl.-Ing. degree in Electrical Engineering from Technische Universitaet Braunschweig, Germany, in 2001. He joined the Institut fuer Nachrichtentechnik (Institute for Communications Technology) of Technische Universitaet Braunschweig in May 2001 as a researcher and PhD student and is with the Department of Electronic Media today. His research activities are in the field of digital modulation and channel coding techniques for terrestrial broadcast systems. He is a member of the DVB Project’s ad-hoc group 'DVB-H', which defined the technical specification for the DVB-H system, and a member of the technical board of the 'DVB-H in Northern Germany' project.

**Performance Comparison of the FLO and DVB-H Mobile Broadcasting Systems**

Khaled Daoud (Braunschweig Technical University, Germany, Germany)

Abstract Mobile broadcasting systems have to overcome a number of challenges, which are mainly related to the characteristics of the handheld devices. These devices have on the one hand a limited battery life time; on the other hand their antennas have a relatively low gain and they are usually used in mobile environments which leads to a poor signal quality. FLO (Forward Link Only) and DVB-H (Digital Video Broadcasting - Handheld) represent two examples of mobile broadcasting technologies being on the verge of commercial deployment. In this paper we will analyze and compare the performance of both systems in a mobile environment based on simulation results. We will focus on the effect of the technologies which distinguish the systems from one another like the carrier interlacing structure, the pilot structure and the channel coding technologies. Index Terms: mobile broadcasting, FLO, DVB-H, interlaces, pilot structure. I. INTRODUCTION Mobile broadcasting systems aim at providing handelds with video, audio and data services. FLO (Forward Link Only) and DVB-H (Digital Video Broadcasting -
Hybrid WiMAX and DVB-H Emulator for Scalable Multiple Descriptions Video Coding Testing

Chee Hock Liew (Centre for Communication Systems Research, United Kingdom); Stewart Worrall (University of Surrey, United Kingdom); Marina Mota (University of Aveiro, Portugal); Antonio Navarro (University of Aveiro, Portugal)

Abstract This paper discusses the design of a wireless emulator for the study of H.264 Scalable Video Coding (SVC) and H.264 based Scalable Multiple Description Coding (SMDC) transmission. This emulator is constructed...
as a part of European SUIT project for examining the benefit of SVC and SMDC for digital TV broadcasting. The individual component of the emulator and some simulation result of wireless bearer (i.e. WIMAX and DVB) will be discussed in this paper. Index Terms—Emulator, WIMAX, DVB-H, TV. I.INTRODUCTION This paper proposes an Hybrid WIMAX and DVB-H emulator (HIWIDEM) for the testing of SVC and SMDC streams over WIMAX and DVB wireless bearer for European Scalable Ultra Fast Interactive TV (SUIT) project (suit.av.it.pt). The overall SUIT architecture is discussed in [1]. Two SMDC descriptions are synchronously streamed from the playout to DVB and WIMAX bearer. A SUIT receiver will be able to listen to both bearers for MDC descriptions. Because WIMAX and DVB-H uses different frequency band, this creates frequency diversity and is exploited by SMDC for robust digital TV transmission. As long as either description is received, the receiver will be able to view the transmitted service (if both description is received, the video quality is enhanced). SUIT architecture details are in [1]. For the first stage of testing, we have considered using an emulator for controlled environment lab testing before the real field trial. The emulator, HIWIDEM, is placed between the playout either the user terminal or the home gateway.

HIWIDEM mimics the behavior of wireless channel and offers the flexibility parameters tuning (e.g. speed, Signal to Noise Ratio (SNR), and Modulation and Coding Scheme (MCS)). HIWIDEM offers a cheap and fast way to test SVC and SMDC in a more controlled way. The following section discussed the emulator in brief. II. HIWIDEM The emulator has a Graphical User Interface (GUI) that is placed in a PC with four Ethernet cards. Each pair of Ethernet card is linked together as an Ethernet bridge for emulation of WIMAX and DVB bearer separately. The main underlying components of the emulator are composed of packet filtering module, WiMAX processing engine and DVB-H processing engine. Packet filtering module can easily split incoming SMDC streams into individual wireless bearer, i.e. WiMAX and DVB-H, since each of the streams is transmitted using different source IP address. The packet processing module utilizes Linux netfilter [2] capability to capture packet from the network. In WiMAX and DVB processing engine, the incoming packets are processed according to the standard [3][4], i.e. packet fragmentation and encapsulation packet payload along with dummy protocol headers to construct smaller source blocks for transmission. At this stage, the source blocks are compared to pre-simulated error pattern files to decide if the transmission is successful. If the transmissions are not successful, the packet that is encapsulated in the source blocks will be dropped while in converse, it is released back into the network. The pre-simulated error patterns are composed of traces of different SNRs for different MCSs generated using physical layer WiMAX and DVB baseband simulator. At 60km/h, 3/4 FEC, 64-QAM, the results show that the minimum SNR to achieve quasi-error free communications over either WIMAX or DVB-H is 25 dB. III. CONCLUSIONS This paper has discussed an emulator that is suitable for the testing of SMDC video transmission over WIMAX and DVB wireless bearer. The emulator, HIWIDEM, will be used subsequently in the project to generate preliminary quality estimation of SMDC transmitted video before field trials. Accompanying discussion of link level simulation is available in the complete version of the paper. The error traces will freely be made available for the research community in the project webpage (suit.av.it.pt) soon. REFERENCES [1] A. Navarro, "SUIT-Scalable, Ultra-fast and Interoperable Interactive Television," submitted to IEEE ISCE 2007. [2] Netfilter Project, http://www.netfilter.org [3] "Part 16: Air Interface for Fixed and Mobile Broadband Wireless Access Systems. Amendments 2: Physical and Medium Access Control Layers for Combined Fixed and Mobile Operation in Licensed Bands and Corrigendum 1," Dec 2005 [4] ETSI EN 300 744, v.1.5.1, "Digital Video Broadcasting (DVB): Framing structure, channel coding and modulation for digital terrestrial television (DVB-T), ETSI standard," Jan 2004. Antonio Navarro (S’89-M’97) graduated (five years first degree) in electrical engineering from Coimbra University, Portugal in 1989 and received the MSc and PhD degrees from the University of Coimbra, Portugal and the University of Newcastle, UK in 1993 and 1996, respectively. He is currently Professor at the Electronics and Telecommunications Engineering Department at Aveiro University, Portugal. In the 2nd semester of 2004, he was on sabbatical leave at University of Southern California-USA. His research interests are on information theory, optimization, rate-distortion, digital television, video coding, video scalability and transcoding, multiple description coding and reliable wireless transmission of video based multimedia services. Antonio has participated and led successfully more than 20 national and European projects and co-authored over 80 papers and one patent. Antonio is the Leader of SUIT, a recently started European IST project (IST-4-028042) in the area of multiple description scalable video coding over DVB-T and WiMAX convergent networks and is now also leading a national project involving an highly computational efficient implementation of an H.264 scalable encoder. Recently, in 2005, he received the IT Award for outstanding scientific achievements. Antonio is the Organizing Committee Chair of the 12th IEEE International Symposium on Consumer Electronics (ISCE 2008). He is currently Associate Editor for IEEE Trans. on Circuits and Systems for Video Technology.

**Next Generation Network ★**

Shrinivas Dharwadkar (MES College Of Engineering, India); Nabegha Masood (MES College Of Engineering, India)

The technological advancements in telecommunication is forcing a trend towards unification of network & services, setting up a stage for the emergence of Next Generation Network-NGN. NGN is essentially an IP based network that enables any category of customers to receive wide range of services such as voice, data & video over the

http://edas.info/showProgram.php?_qf__showProgram=&c=5136&form...bio%5D=1&program_view=everybody&action=Show+conference+program
same network. The service layer in NGN is independent of underlying network and access is enabled across a wide range of broadband technologies, both wireless such as 3G, Wi-Fi, WiMax and wire line such as Copper DSL, cable, fiber etc. The different transport networks are replaced by a single IP network. The Migration to NGN reduces network and operational complexity resulting in better & reliable service. It offers unrestricted access by users to different service provider .It supports generalized mobility, which will consistent provision of services to users. In migration to NGN security is crucial. In IP based, which is convergent and has open environment, network infrastructure must be flexible and open towards various types of service. The key technique for NGN is scalable video coding and QoS to adapt to the various requirements such as quality, spatial and temporal resolution and bandwidth variation in heterogeneous networks. NGN would employ a meshed core, having embedded intelligence which would provide scalability, throughput and enhanced revenue generation by providing optimized connection, service, flexibility and efficient network management thus providing class of service (CoS). The constant human endeavor to communicate more effectively without any physical & psychological bondage has led to the trend of evolution in telecommunication –NGN.

Wireless Sensor Networks 1

Room: Trinity 4

A Body Sensor Network Platform with Two-Level Communications

Dongheui Yun (Information and Communications University, Korea); Jihoon Kang (Information and Communications University, Korea); Jae-Eon Kim (Information and Communications University, Korea); Daeyoung Kim (Information and Communications University, Korea)

Wireless Sensor Network (WSN) technologies have been extended to the bio-medical area, and it is called Body Sensor Networks (BSN). BSN systems sense and transmit the vital signs of human, such as electrocardiogram (ECG) and electromyogram (EMG), in unobtrusive and efficient way. Those vital signs are critical to human’s life and behavior, so the data should be reliable and transmitted in real-time. In this paper, we propose a BSN platform that is using two-level communications (TLC) to increase the reliability of the system. Also, we develop a hardware and software platform to support the TLC and increase the reliability and energy efficiency.

Proactive Services using RFID and Sensor Network

Young Cheol Go (ETRI, Korea); Minsu Jang (ETRI, Korea); JooChan Sohn (ETRI, Korea); Hyun Kyu Cho (ETRI, Korea); Young-Jo Cho (ETRI, Korea)

This paper aims at providing proactive service for ubiquitous robot and ambient intelligence. Before long, a robot will be considered as home appliances. But unlike usual home appliances, a robot can move and do services with and without actions. A robot can assist people physically in their daily life and live together with them like a member of family. So users would expect robots to do different services from those of home appliances. In the true ubiquitous environment, people can be served without noticing the presence of service providers. But it is difficult for current robots to provide services without the interaction of users. By the nature of the robot service, there are services in which the interaction with a man is needed. However, there are also services provided without user's request. In this paper, we focus on the method for providing the later service. Through properly responding to the command of a man, robots can give convenience to people. In the situation without the specific command, if the appropriate service can be provided for a man, the benefit of a robot will become much higher. But it is difficult for robots to do the right service without user's request while satisfying service recipients. Such difficulties are as follows. First, if robots do something without interactions they have to know the various kinds of situations in the world and can decide what and how to do from the information they get. Second, the performance and the kinds of current sensing devices are limited. And last, there is the difficulty of the stable environmental information acquisition from sensor networks. In this paper, we introduce a proactive service framework using RFID and sensor networks to overcome the above problem. First, a probabilistic service model is introduced to describe each service situation. We make service models using Bayesian Networks. The service model describes service conditions, service goals, and etc. ‘Bayesian Networks’ is one of the probabilistic reasoning methods showing the stable performance in the uncertain situation. So using Bayesian Networks, it’s possible to overcome unstable sensor data acquisition. Robots provide services under service models. Second, we build a sensor network to catch a user need and decide the right service start point. To get proper input data, if necessary, sensing data are fused and synthesized before use. RFID is also used in order to distinguish identities of people and objects. And last, the process of providing a service in a proactive manner is as follows. We identify three stages of providing a proactive service: attention, motivation, and deliberation. Attention is the activity of monitoring the changing situations and deciding what aspects of situations to pay attention. Motivation is the activity of fuelling the intention to do something. It’s not the result of deliberative reasoning but the result of a mechanism similar to conditional reflex. The motive of a service is generated very fast by the result of the attention process. Deliberation
is the process of understanding and guessing the situation more thoroughly, and elaborating the motivation into a series of concrete services.

In 1997, he joined Electronics and Telecommunications Research Institute (ETRI), Daejeon, Korea, where he is working on the development of the URC Server Framework for Proactive Robotic Services Research Interests - Intelligent Robot, Edutainment Robot, Human-Robot Interaction

**Implementation and Analysis of New Lightweight Block Cipher Suitable for Wireless Sensor Networks**

Woo Kwon Koo (Korea University, Korea)

Sensor devices have critical resource constraints such as processing speed, memory size and energy supply. Especially, energy consumption affects the network lifetime so that energy efficiency is an important requirement for wireless sensor networks (WSNs). It means that it is a considerable matter to choose the energy- and memory-efficient block cipher suitable for wireless sensor networks. TinySec, de facto security architecture for wireless sensor networks, supports traditional block ciphers such as RC5 and Skipjack while the traditional block ciphers might be unsuitable for 8-bit computing devices of which wireless sensor networks consist. Accordingly, it is necessary to evaluate the traditional block ciphers and 8-bit oriented block cipher in performance but there is no work in this area. In this paper, we consider another candidate HIGHT, designed to be proper to ubiquitous 8-bit computing devices (e.g. sensor node or RFID tag), for wireless sensor networks. After implementing new lightweight HIGHT on MICA2 and analyzing the performance between HIGHT and the traditional block ciphers, we can conclude that HIGHT is more suitable for TinySec than the traditional block ciphers. Hence, we recommend new lightweight candidate HIGHT to replace RC5 and Skipjack as security module in TinySec.

**Authenticated Group Key Distribution for Wireless Sensor Networks**

Hwaseong Lee (Korea University, Korea)

Wireless sensor networks are utilized in various applications and also have the high potential to be developed in near future. To guarantee secure communication in wireless sensor networks, secret keys should be securely established among sensor nodes. Recently, Chadha et al. proposed a group key scheme for wireless sensor networks via local collaboration. A group key was derived from base station’s broadcast message and a node’s secret. However, there is no authentication in the group key scheme. It is essential to share a reliable group key among the entire nodes even though it results in a little overhead. Our scheme strengthens a group key distribution of Chadha et al. and can be applied into other schemes.

**3:15 PM - 5:00 PM**

**Consumer Networks 7**

**Dynamic Contention Slot Allocation for 802.16 Broadband Wireless Access Systems**

Yanling Yao (STMicroelectronics, P.R. China); Hongfei ZHU (STMicroelectronics, P.R. China); Jonny SUN (STMicroelectronics, P.R. China)

The IEEE 802.16 standard for Broadband Wireless Access (BWA) is promising to offer high-speed wireless connectivity between Base Station (BS) and Subscriber Stations (SSs). The wireless link between BS and SSs is a two-way shared medium, where the downlink is a broadcast channel exclusively used by BS and the uplink channel is shared by all SSs on a demand basis. In each frame, BS allocates a number of Contention Slots (CSs) for SSs to send bandwidth reservation request through contention. This allocation of CS will directly affect the system throughput. However, how to calculate the number of CS is left open by the standard in terms of implementation. In this paper, we analyze the impact of Contention Slots Allocation (CSA) on system throughput and thus propose an algorithm to optimize the utilization of uplink bandwidth by dynamically adjust the number of CSs. Analytical and simulation results show that the proposed CSA algorithm can efficiently improve throughput for 802.16 systems.

Hongfei ZHU got her Master's Degree on Communication & Information System field from Peking University in 2005. After graduation, Hongfei joined STM in Beijing, in Broadband Wireless MAN group as a R&D system engineer. Last year, her paper on QoS scheduling issue was published on VTC spring conference.

**RFID-Based Desktop Migration System**

Taein Hwang (ETRI, Korea)

Mobile users constantly need to create and personalize their working environments once they move from one
Design and Implementation of ZigBee based Universal Remote Control (URC) Applicable to Legacy Home Appliances

Wan-Ki Park (ETRI, Korea); Chang-Sic Choi (ETRI, Korea); Jinsoo Han (ETRI, Korea); Intak Han (Electronics and Telecommunications Research Institute, Korea)

Many various home appliances, most of which are controlled by IR (Infra-Red) signal based remote controller, are used in today's digital home environments. A universal remote control (URC) unit is an integrated device for controlling many different consumer electronics (CE) home appliances with a single one. To control these various CE devices based on IR (infra-red) control signal, the URC unit has to have many preprogrammed control codes for controlling them because they are controlled with many different types of IR profile stated with lead code, control code, carrier frequency, duty cycle, duration and so forth. In addition to this difficulty, the IR based URC can control any devices under the only condition of line of sight. In this paper, we propose a dynamic control scheme for IR controllable multiple legacy CE appliances, which is based on IEEE 802.15.4, especially ZigBee protocol. The proposed scheme uses two types of main components. One is a URC unit using ZigBee based wireless network technology, WPAN (wireless personal area network). Another is a Z2IR (ZigBee to IR) conversion module which converts a control message transferred through the ZigBee network into an IR typed control signal. The ZigBee based URC, called as Z-URC (ZigBee based universal remote control), operates as a ZigBee coordinator in our proposed scheme, while Z2IR modules, which are attached to target appliances, operate as...
ZigBee devices. In the proposed scheme, the list of CE devices to be controlled by the Z-URC is dynamically reconfigured. It is based on his location as a consumer with the Z-URC goes from a room to any other place in his house. That is to say, the Z-URC shows a list of CE devices to be controllable with it. The proposed Z2IR contains only a specific IR control codes for its target appliance, which codes is initialized and installed to the Z2IR module during setup procedure. A user may configure its product and vendor identifiers (IDs) of the home appliance to be controlled by the IR control signal from the module. Based on this feature, the module informs Z-URC target appliance’s information and takes its IR control codes for its target appliance through ZigBee WPAN. And, we apply a novel mechanism to minimize power consumption of the Z2IR module, which is based on ON/OFF mechanism of ZigBee devices using multiple timers. This paper specifies in detail the proposed remote control scheme and implementation of hardware, firmware and software for the proposed ZigBee based URC mechanism applicable to legacy IR controllable CE devices.

**Smart Adaptive Remote Controller, A New Method for Consumer Electronic Equipments**

Ahmet Ozturk (BEKO Electronic, Turkey)

Abstract—This document is briefly explain a new concept about remote controllers. This concept is depending on a buttonless design and drawing keys on a touch screen each state of the controllable device. Continuously growing technology and changes in customer needs have brought together a vast increase in the number of consumer electronic devices. With each new device, new features are introduced in parallel to this growing technology. New features mean more buttons on the devices’ remote controllers. Remote controllers were unable to show the same technological improvements as the new technology consumer electronic devices. Remote controller concepts in use today enable us to design only controllers with fixed number of buttons. Nevertheless some remote controllers that exist in the market today can define their own buttons and even macros by their properties or by aid of the computers. But all remote controllers basically control the devices with single directional communication by sending key codes with the help of protocols they use. To realize the same technological development achieved in consumer electronic devices a new concept is needed for their remote controllers. This approach is based on the principle of changing the number of keys on the remote controller that can be used according to the state of the controlled device. The aim of this paper is introducing a new remote controller approach to develop more flexible applications and provide a framework for new kind of applications. In this approach bidirectional communication is targeted. The main aim of bidirectional communication is reducing or limiting the number of keys. Thus users will deal with smaller number of keys consequently the system will be less likely to crash on any given condition as the user will not be able to press unsuitable keys. In order to provide bidirectional communication we need more than infrared medium like radio frequency systems. The most suitable radio frequency system is wireless networks. Because these systems are started to already use some consumer electronic products and very reliable systems. Also bluetooth is another alternative radio frequency system to realize this new approach. To make this approach possible a new well defined protocol is needed. Remote controllers enable the users to control all devices that are compatible with this concept and protocol. In order to provide this, there must be a common protocol. Thus users will not deal with many remote controllers and will not keep many remote controllers in hand to control any device when needed. In addition to this, consumer electronic producers can design generic remote controller for their product by supporting specific protocol. If we make a list why this new remote controller approach is needed we can see the benefits of this system. • User deals with limited and usable keys every time. Users always see the keys that related with any system’s state. • Remote controllers can be control almost every device that support the protocol. • Devices are not crashed by user by pressing an invalid key while state changes of the system. • Possible to display keys which are mostly used in any system states by learning user’s habits with an artificial intelligence technique. In this abstract explained that why consumer electronic products need an adaptive and universal controller and this type remote controller’s structure and its protocol. The protocol which is called Smart Adaptive Remote Controller Protocol and other mechanism are defined in or study. The protocol provides a common language to make the remote controller and the controllable devices understand each other. In order to prove this smart adaptive remote controller protocol applicable, a simulation program is developed.

EDUCATION


**Dynamic Sensing Level Selection for Energy Efficiency in Wireless Sensor Networks**

Seongwoo Kim (Korea University, Korea)

In this paper, we consider a dynamic sensing level selection and energy limitation problem for wireless sensor networks. Accurate, reliable and timely sensing information needs more frequent sensing. In other words, energy is necessary for sensing operation. This implies a trade-off between sensing rate and energy consumption. In this paper, we propose the practical energy-sensitive approach exploiting such a trade-off, where sensing rate can be changed dynamically depending on coverage statistics and event detection utility. We formulate energy efficient
problem under Markovian assumptions, where the objective is to provide a certain degree of coverage capacity and probability so as to maximize the network lifetime. We provide the mathematical modeling of the dynamic sensing level selection, and an algorithm for exploiting the trade-off between energy and sensing level. Finally, we derive the boundary issue of our suggestion.

Mobile Computing & Communication 5

Room: Trinity 6-7

Design and Implementation of a GSM Payphone
Reza Anvari (University of Tafresh, Iran); Mehrdad Nourani (University of Texas at Dallas, USA)
This paper presents design and implementation of a GSM payphone. The design integrates two micro controllers, GSM engine and some analog and digital parts. Detailed description and implementation of each design element are presented.

HySEC: a Hybrid Scheme to Enforce Cooperation in Mobile Ad hoc Networks
Saeed Soltanali (Amirkabir University of Technology, Iran); Sajjad Pirahesh (Amirkabir University of Technology, Iran); Salman Niksefat (Amirkabir University of Technology, Iran); Masoud Sabaei (Amirkabir University of Technology, Iran)
A Mobile Ad hoc Network (MANET) is a set of wireless mobile nodes that dynamically function as a network without relying on any preexisting infrastructure and centralized administration. It is an autonomous system where each node operates not only as an end system but also as a router to forward packets for other nodes. Communication between two distant nodes relies on cooperation of intermediate nodes to forward packets for the benefit of sender and receiver. Cooperation means that nodes perform network functions for benefit of other nodes. In most current ad hoc routing protocols, it is assumed that all nodes in the network follow the routing protocol and are willing to forward packets for other nodes. But a selfish node, in order to save its resources (e.g. battery power, bandwidth) does not cooperate and drops packets which are not of its direct interest while still using network to relay its own traffic. If there is no mechanism to cope with selfish nodes, number of selfish nodes will increase and cooperative nodes may find themselves unfairly loaded, thereby network performance degrades extremely. Generally cooperation enforcement schemes in MANETs could be classified as Virtual currency based schemes and Reputation based schemes. In this paper we propose a scheme to counter with selfish nodes and enforce cooperation in mobile ad hoc networks. Our proposed scheme is a combination of both type of aforementioned approaches and mitigates disadvantages of them. Proposed scheme is based on a fully distributed reputation mechanism in which each node monitors its neighbors (i.e. one hop distant nodes) and computes a reputation rating for them, based on observed behavior. To improve rating accuracy and cognition speed, each node also regards its neighbor's opinions about other nodes in calculation of reputation values. In fact, a node's reputation indicates the reliability to its forwarding functionality. Each node uses reputation rating beside another parameter called Credence Factor that represents credence of a node to its assessment about other nodes, as metrics to judge about selfishness and reliability of other nodes. For punishing selfish nodes, each node only forwards packets of non-selfish nodes. Proposed scheme don’t need central control or a priori trust between network nodes. Each node individually judge about other nodes selfishness and a consensus between nodes is not necessary. Another important parameter which is worked out in this scheme is Debit counter. Each node keeps its account balances with other nodes to limit its services to other nodes proportional to their reputation rates. In other words, this parameter adjusts an upper-bound for packet forwarding for each node. Each node uses its reputation information to choose most reliable shortest path for sending its packets. Applying this strategy increases the probability of packet reception to the destination. GloMoSim is used to simulate proposed scheme. We implemented our scheme as an extension on DSR routing protocol. Simulation results represent that throughput of well-behavior node improves up to 35% when as 40% of nodes are selfish. Furthermore, selfish nodes are detected in 82% accuracy by nodes which apply this scheme. Proposed scheme imposes 14% traffic overload to the network.

DAP: Dynamic Address Assignment Protocol in Mobile Ad-hoc Networks
Haeyong Kim (Electronics and Telecommunications Research Institute, Korea); Sang Cheol Kim (Electronics and Telecommunications Research Institute, Korea); Misun Yu (Electronics and Telecommunications Research Institute, Korea); JunKeun Song (Electronics and Telecommunications Research Institute, Korea); PyeongSoo Mah (ETRI, Korea)
In general IP networks, addresses can be assigned to hosts manually by a network administrator or automatically by a DHCP server. Mobile ad-hoc network (MANET), which is a self-configuring network, is the union of mobile hosts that form an arbitrary topology. Most research related to the MANET assumes that host's IP address is...
configured, prior to the node joining the MANET. However, it is impossible that a network administrator or DHCP server configures host IP addresses in MANETs because there is no infra-structure. For this reason, a dynamic address management protocol is essential in MANETs. Thus, this paper proposes a novel self-configuring address management protocol, referred to as DAP (Distributed Address Pool), in which IP addresses can be dynamically allocated to a newly-joined host in MANETs with no network infra-structure. In DAP, every host in a MANET has a unique IP address pool (a set of unused addresses that will be used for new joined hosts in MANETs) and address assignments are performed locally in the host. The advantage of this method is that it does not generate any broadcasting messages and, in turn, address allocation time can be significantly reduced. From our simulations, DAP showed a superior performance to a random address allocation scheme (RADA) in terms of both address allocation time and message exchange overhead.

**DLP Projection: Migration to Ultra-Compact Platforms**

Tom Winter (Texas Instruments, USA); Brian Dodge (Texas Instruments, USA); Dan Morgan (Texas Instruments, USA)

DLP technology has moved from an expensive novelty to capturing 50% of the front projection market (and 100% of the digital cinema market) in the 10 years since first product introduction. Business projector form factors have evolved from the first 50 lb behemoth (at 400 lumens) to mainstream 1,500 lumen projectors less than 5 lbs. New emerging markets in 2006 includes instant theatre (integrated DVD player, speakers and projector), and pocket projection. "Pocket" projectors are a new category of <1 lb units with solid state illumination (LED or laser), starting at 50 lumens in 2006 and moving up to 200+ lumens in 2007. New packaging technologies (including wafer level packaging) will allow these units to progress to 1" thickness and substantially reduce chipset and other component costs. The next logical extension of ultra-portable projectors is tapping into the multimedia expansion of mobile handsets. Integration of a multimedia cell phone with a docking station projector would have obvious applications as more and more handsets are "mobile TV" compatible. Design considerations for full integration of a laser based projection unit into a cell phone will be discussed, as well as a demo of a prototype cell phone projector.

**Design of Wearable Gadgets for Life-Log Service based on UTC**

DongWan Ryoo (ETRI, Korea); Jong Ho Won (ETRI, Korea); Chang Seok Bae (ETRI, Korea)

According to the appearance of very large capacity memory device, the many empirical information or interesting events of an individual can be recorded. The new demand to store user's special situations or the daily life appears accordingly. There have been some works to log person's empirical information or the daily life in the area of mobile computing and wearable computing. A life-log means a recording the empirical information or the daily life of an individual as the digital method. The activity, a photograph, physiological signals, time information, and location of user are can be included in life-log. Recently, the decline of the digital storage cost makes it possible to store this information in the digital media. The intelligent gadget means that the system has the abilities of the information gathering, the information processing, and the low power communications. Moreover, the system can be used to the individualized service. A user can store the empirical information in a life-log system by using these intelligent gadgets. A daily-life information of an individual such as video, audio, location, physiological information, and etc can be acquired and stored by the intelligent gadget. The life-log system means that the system anytime and anywhere can collects the everyday information of an individual and can records and can search the information. In this paper, the design of wearable gadgets for life-log service based on UTC is presented. We designed of two kinds of wearable gadgets. The firstly is wearable gadget for low rate data (e.g., location), the second is wearable gadget for high rate data (e.g., video). Generally, the generated data from the different devices are not included the exact acquisition time. It is difficult to realize the exact acquisition time of the generated data from the different devices. Therefore, the data generated from the different electronic device could not be matched by synchronization. For example, In case of the signals coming from sensors connected to the same system, these signals use the same system clock. Therefore the signals can be combined and can be merged. But it is difficult to make the synchronized sensing data with the data generated from the different electronic device. The gadgets acquire the time information (UTC) by using GPS from a satellite. The acquisition time information (UTC) can be included in the acquired data coming from the different systems such as an image, an audio, a bio-signal, environmental information, etc. Also, the measured life-log signals and environment signals in a daily life can be merged with the acquired time and position information. Therefore it can makes synchronization between generated data from the different apparatus and we can get the more available information. This information can be realized the obtained time of life log data later. It makes more and more useful information can be obtained. This can also provide many usabilities to retrieve data of life log.

Dong-Wan Ryoo is with the Department of Post-PC Research Group Digital Home Research Division, Electronics and Telecommunications Research Institute.

**Wireless Sensor Networks 2**
Room: Trinity 4

**A2S: Automated Agriculture System based on WSN**

Seong-eun Yoo (Information and Communications University, Korea); Jae-Eon Kim (Information and Communications University, Korea); Taehong Kim (Information and Communications University, Korea); Sungjin Ahn (Agency for Defense Development, Korea); Jongwoo Sung (Information and Communications University, Korea); Daeyoung Kim (Information and Communications University, Korea)

This paper describes the results of 1-month deployment of A2S which consists of 3-sub sensor networks and a management sub-system to manage the WSNs and to provide various and convenient services to consumers living in a farming village. 25 sensor nodes, 1 actuator node, and 3 sink nodes were deployed in greenhouses with melon and cabbage and operated during severely cold winter and one of the sensor nodes was plunged to a field near to one of greenhouses in order to endure very harsh condition of heavy snows and coldness of -15 celssius degree. 3 industrial PC-based gateway were installed in the greenhouses to provide WLAN links between WSNs and management sub-system which was in a room of Dongbu Handong Seed Research Center about 0.5km far away from the greenhouses. The management subsystem with a DB-server and a web-server manages WSNs and provides easy interface to farmers with hand-help devices such as a PDA. We learned valuable ideas and experiences from this real deployment and operation of A2S and believe they can be useful in consumer electronics field such as home network as well as automated agriculture field. We present A2S from A-node(Agriculture sensor Node) hardware and software to the overall architecture of A2S.

**NanoMon: An Adaptable Sensor Network Monitoring Software**

Misun Yu (Electronics and Telecommunications Research Institute, Korea); Haeyong Kim (Electronics and Telecommunications Research Institute, Korea); PyeongSoo Mah (ETRI, Korea)

A wireless sensor network (WSN) is a collection of sensor networked devices carrying information around the physical environment. There are a lot of WSN applications such as environmental monitoring, home automation, object tracking, security and prevention of disasters. In those applications, tens or hundreds of sensing devices generate a large volume of data. To deal with the management issue of the large volume of data, a monitoring software with decent abilities is essential. In this paper, we present a sensor network monitoring software, named NanoMon, which has a flexible architecture and supports for various user requirements of sensor network applications in an adaptive manner. With NanoMon, users can specify custom GUI plug-ins and internal module parameters by using a simply describable configuration file; and it can be automatically integrated to NanoMon framework to support user-specific sensor network applications. NanoMon employs a widely used database, MySQL, to concurrently and correctly manage sensing data and node information of several types of sensor network applications. To show flexibility and adaptability of NanoMon, we implemented two WSN applications – home circumstance monitoring and parking lot monitoring systems. By the selected application name, NanoMon easily changed its GUI and internal module parameters such as sensor types and sensor data units used to display the status of WSN using the configuration file described by users.

**Embedded Semantic Engine for Ubiquitous**

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We present in this paper the design and implementation of the so-called “embedded semantic engine (ESE)” with experimental usages of the engine on ubiquitous appliances including a mobile surveillance robot and a PDA-based healthcare terminal. ESE is an embedded software platform for context awareness and intelligent decision making. It basically consists of four main components: a context interpreter, a situation board, a service manager, and an inference engine. The context interpreter is interfaced to the sensor networks from which various sensor data packets are collected. The interpreter abstracts the sensory data into a set of symbols and puts them into the situation board. The situation board dynamically manages and updates the interpreted contexts according to the pre-specified context management policies, which can preserve the correctness of the situation that is presumed by the content of the board. The service manager determines which control commands to be transferred to the controllers. For robots, the commands determine the target spot to navigate, some sound waves to emit, etc. For the health-care terminal, the commands alter various health-status indicators. The core of ESE is the inference engine. The inference engine is used by all the other components to interpret sensor data, manage contexts, and determine control commands. The inference engine for ESE is basically a rule engine with a capability to perform inference over RDF and OWL documents. RDF and OWL are W3C recommended languages for representing semantic data on the web. So, ESE-enabled ubiquitous appliances can exchange data with RDF/OWL compliant peers including web services that can understand and provide RDF/OWL data. Also, it should be noted that we can put meaning to the data with RDF/OWL. Based on the capability to process RDF/OWL, enhanced services are possible as exemplified by the case of our implementation of a sensor network and robot based surveillance system, which are summarized as follows: - A surveillance robot with ESE can discover and accept a set of intelligent surveillance policies written in OWL from a trusted policy provider on the web, and instantly apply the
policies for its surveillance service. - A robot can perform surveillance services for previously unknown spaces by downloading and interpreting the space maps written in RDF or OWL. - We can describe specific regions on the space map with some meaningful symbols to derive more intelligent services from robots. In our implementation, we marked specific regions as restricted areas so that they can receive more attention from the surveillance robot. If map builders share the symbols, then the robot would exhibit similar intelligent actions for the space maps built by different map builders, which would elevate the value of the robot. - ESE enables automated service deployment for newly introduced sensors. In our implementation, if a new sensor is introduced into the environment and powered on, it emits a heartbeat signal with its unique ID. ESE accepts the unique ID, retrieves the sensor capability profile based on the ID, and initiates a service discovery session to find services that are supported by the sensor. Once services are discovered, the services can be deployed automatically or selectively by the user. This automated service deployment is found to be crucial for lowering the cost of initial installation of our robotic surveillance system as well as ongoing maintenance. We developed two versions of ESE/Java which is in Java, and ESE/C++ which is in C++. The size of each ESE version is about 700KB total, with low footprint for moderate input data. So, ESE can be utilized on various embedded platforms including machines with embedded linux as well as any J2ME enabled machines. Based on ESE's inferencing and context-awareness capabilities, it's possible to build small-sized embedded intelligence applications for ubiquitous appliances. ESE-based intelligence applications can enhance the data processing and service provision capabilities of the appliances. Also, the connectivity of the appliances and the data interoperability provided by ESE would open up a possibilities of building collaborative and community-based applications for ubiquitous appliances.

A Dynamic Stack Allocating Method in Multi-Threaded Operating Systems for Wireless Sensor Network Platforms

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Typical sensor nodes have a small amount of memory with 2-10 KB and even no hardware devices for memory protection such as MMU. Consider multi-threaded sensor applications running on such a memory-constrained hardware platform. In most sensor operating systems, it is assumed that thread stacks are statically allocated. However, this static allocation is not appropriate for memory constrained sensor hardware because shortage of stack memory space can bring in the stack overflow problem. As an alternative method, this paper proposes a dynamic stack allocating method, which enables to adaptively adjust the stack size of each threads based on the stack usage information. The information of the stack usage is obtained by measurement at run-time. The proposed method also defines a stack reallocating problem and solves it in n steps, where n is the number of thread stacks. Our experimental results showed that the proposed method significantly minimizes the waste of thread stack memory space compared to the static stack allocating method.