The AES 29th Conference, *Audio for Mobile and Handheld Devices*, was held at the conference center (see photo inset) on the beautiful campus of Seoul National University located in the thriving metropolis (background photo) of Seoul, Korea on September 2–4 2006. The conference was a significant milestone for AES. It was the first AES conference held in Asia. It was also the first conference devoted to audio in mobile and handheld systems, which is now becoming a significant sector of the audio and music industries. This three-day conference brought together researchers and developers in the fields of both audio engineering and mobile telecommunications.
Twenty four papers were presented including a keynote talk, two plenary talks, and four poster presentations. There was also a workshop on power-efficient audio. Excellent papers were presented covering most of the topics related to handheld audio. The papers covered topics such as coding, speech processing, 3-D and synthetic audio, Class-D amplification, and various implementation issues. Six companies demonstrated their products throughout the conference. A total of 82 participants attended from 15 different countries.

SATURDAY SESSIONS
The conference opened on Friday morning with an introduction from Conference Chair John Oh of Pulsus Technologies. Koeng-Mo Sung, of Seoul National University and president of the Institute of Electronics Engineers of Korea, gave a welcoming address. Jung-Hee Song, of the Korean Ministry of Information and Communication, explained the efforts of the Korean government concerning advanced information services related to mobile audio.

The academic program started with the keynote speech of Won-Yong Jo of SK Telecom, entitled “Perspective of Mobile Music Service.” SK Telecom is the biggest mobile telecommunication service provider of Korea. The company has launched a very successful mobile music service over the CDMA (code division multiple access) mobile networks. Jo has spearheaded the SK Telecom MelOn music service team. The group is now extending this service to...

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other countries; for a low monthly fee subscribers get access to a huge database of music that can be played on PCs, mobile phones, and MP3 players. Different from the Internet download services popular in the U.S. and Europe, this service successfully distributes music over wireless telecommunication networks.

The technical sessions opened with the invited paper, “Multichannel Goes Mobile; MPEG Surround Binaural Rendering,” presented by Jeroen Breebaart of Philips Research. With the proliferation of multimedia phones and personal media players that have limited data-storage space, bit-rate-efficient transmission of high-quality multichannel content over low-speed networks has become an important issue. In this paper Breebaart and his coauthors outline a significant addition to the MPEG Surround technology that enables computationally-efficient decoding of MPEG Surround data into binaural stereo, as is appropriate for mobile devices.

Papers on audio coding were presented in the first session on Saturday morning. Miyoung Kim, Samsung Advanced Institute of Technology, presented the paper “Bandwidth Extension for Scalable Audio Coding.” She described how MPEG-4 BSAC has fine-grain scalability, in which the bitstream can be truncated and decoded at any layer from one full bitstream, but the decoded output loses its high-frequency signals and the sound quality becomes degraded. Kim and her coauthors propose a novel method for recovering the missing frequency signals when the decoded bitrate is lower than the top bitrate.

Geun-Bae Song, Samsung Electronics, presented the paper “Preprocessing Method for Enhancing Digital Audio Quality in Speech Communication System.” This preprocessing method redistributes the spectral energy of the music input all over the spectral domain so that the preprocessed music can be coded more effectively.

Dalwon Jang, Korea Advanced Institute of Science and Technology, presented the paper “Automatic Commercial Monitoring for TV Broadcasting Using Audio Fingerprinting.” Jang and his coauthors determined that video information is necessary to complete such a monitoring system.

The most popular topic of this conference was Class-D amplification. Because of its intrinsic advantage of low power consumption and digital interfaces, this technology should be used as a mainstream technology in mobile and handheld devices.

In the afternoon session, John Oh of Pulsus Technologies presented an invited talk on the digital Class-D amplifier (in other words, full digital amplifier). Full digital is advantageous because it can be used to implement a better quality audio with integrated DSP features, and the digital interface is immune to the problems due to the noisy environment of handheld systems such as mobile phones. However, this technology was only recently adopted in handheld audio systems because of several technological challenges. Oh reviewed the issues of full digital amplification in the handheld environment and the implementation of SoC (System-on-a-Chip) products that can be used to play music on mobile phones.

Vladislav Shimanskiy, Samsung Electronics, presented the paper “Chaotic Modulation in PWM Digital Amplifier.” He used noise-like phase modulation of PWM carrier signal that allows a reduction of nonlinear effects in the demodulation filter and can be advantageous for EMC reasons. He proposes a method of digital PWM amplifier implementation in which a digital chaotic oscillator is used for carrier spectrum spreading.
Gaël Pillonnet, ST Microelectronics, in “A Hybrid System Approach for Class-D Audio Amplifier,” presented a new hybrid control model for Class-D analogous to that of motor drives and dc-dc conversion.

Pascal Tournier, On Semiconductor, presented an evaluation of Class AB versus Class-D amplifiers for multimedia applications in portable electronic devices.

There was a short poster session during the afternoon break that gave attendees an opportunity to ask in-depth questions of the presenters. The papers presented during the poster session were “Dual Channel Audio Decoding Architecture of Digital TV SoC” by Hyo Jin Kim et al. of LG Electronics; “Multichannel Sound Scene Control for MPEG Surround” by Seungkwan Beck et al. of the Electronics and Telecommunications Research Institute of Korea; and “Evaluation of PEAQ for the Quality Measurement of Perceptual Audio Encoders” by Ashok Magadum of Ittiam Systems.

Late in the afternoon on Saturday the conference planners offered an interesting addition to the technical program. Entitled Industrial Solutions, the session was favorably received by the attendees. Representatives of the official conference sponsors—Samsung Advanced Institute of Technology, LG Electronics, Pulsus Technologies, and Oxford Digital—in addition to On Semiconductor and Fraunhofer IIS, gave brief overviews to the products that they would be demonstrating during the break periods throughout the three days of the conference. The information that attendees received in this session gave them an opportunity to pencil in time in their schedules for follow-up questions at the demo stands.

SUNDAY TECHNICAL SESSIONS

The two sessions on Sunday morning looked at implementations and speech processing. Andreas Ehret, Coding Technology, discussed the implementation of color ring-back tone service using the aacPlus codec. Current ring-back tone service uses low-quality audio codecs. By using aacPlus, providers can offer higher quality service to subscribers. For instance, the traditional beeps you hear when you wait for the person you are calling to open his phone and begin your conversation can be replaced with music to calm your nerves.

Manish Arora, Samsung Electronics, presented the paper “Low Complexity Virtual Bass Enhancement Algorithm for Portable Multimedia Device.” This algorithm uses simple real space implementation that can be easily utilized with low computational complexity.

Hyun-O Oh, LG Electronics, presented a fast quantization loop algorithm for MP3/AAC encoders. Encoders are still hard to implement on power-limited systems as they need much higher computational complexity and memory. Low-power, high-quality encoders are required for convergence in broadcasting and communications.
Martin Schönele, Siemens Research, introduced a state-of-the-art hands-free audio system for mobile phones, which has such features as acoustic echo cancellation, noise reduction, speech-intelligibility enhancement, and wideband audio extensions.

Naofumi Aoki, Hokkaido University, presented a paper on a bandwidth extension technique for the G.711 speech codec. He proposed a new gain-adjustment method to avoid problems of conventional techniques in the case of signals with larger spectral components in the high-frequency range above the upper-bound frequency of the codec. By transferring the different coefficient of band extension varying according to the frequency, he was able to get a better reproduction of tones such as the “sh” sound.

Yang-Won Jung, LG Electronics, discussed an adaptive microphone array technique with self-delay estimator. He proposed the technique as a promising method to restore clean speech signals in noisy environments. Reverberation is one of the main difficulties for existing systems that affect the performance of the algorithm. To avoid using a separate delay-estimation module, he has introduced an adaptive blocking matrix that can self-estimate delay.

A MEMORABLE CULTURAL EXCURSION

After the morning technical session, attendees spent the rest of the day visiting some of Seoul’s most important cultural sites. First they fortified themselves for the journey with lunch at a restaurant famous for its ginseng chicken soup.

The next stop was a guided tour of the National Folk Museum of Korea, where they learned about the the history and culture of the Korean people.

A short walk from the museum took them to beautiful Kyungbokgung Palace, which used to be the main residence of the emperor of the Chosun dynasty. After enjoying the majestic architecture of the palace, they then went for a short visit to the Jogyesa Buddhist temple.

By this time all the walking and touring had depleted the nourishment of that lunch of ginseng chicken soup. The group was ready for the best part of the tour: an exotic Korean banquet followed by traditional Korean folk music and dance performances at Korea House. The group marvelled at the artistry of the performers, who have been designated “national human treasures” by the Korean government. A number of dances and performance styles were performed, the group’s favorites were the fan dance (see photo on page 73) and the drum performance.

MONDAY SESSIONS

Monday morning began with a second session on implementation. It opened with a presentation by Juha Backman, Nokia, describing collaborative efforts on the audio interface specification “Slimbus” being developed by the MIPI (Mobile Industry Processor Interface) Alliance. The purpose of this new standard effort is to help cellular industries to develop more capable interface specifications for various functions in the handset. This approach defines a 2-wire, multidrop TDMA bus where data communication and control are integrated. While optimized for 48-KHz sample rate isochronous streaming digital audio transfers, it is capable of supporting all common digital audio sample rates.

Hosei Matsuoka, NTT DoCoMo, presented a method of aerial acoustic communication in which data is modulated using OFDM (orthogonal frequency division multiplexing) and embedded in regular audio material. Since many audio devices are located in public spaces, this method can be an effective way of establishing ubiquitous communication. A typical application could be embedding a URL in the audio stream of a TV or radio program; users with mobile handsets could get to that show’s page on the Internet without manually typing in the URL.

Pierre-Louis Bossart, Freescale Semiconductors, discussed mobile architecture issues in his presentation. By reviewing
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the complete mobile audio framework, from audio codecs to audio middleware, from platform architecture to hardware, he provided a succinct explanation of the critical issues faced by the makers of complex multimedia handsets.

The next session was devoted to 3-D audio and synthetic audio. Youngchul Park, Yonsei University, discussed work on low-complexity algorithms suitable for mobile application. He explained that these methods could achieve 50 percent savings in computational power and memory compared to the conventional method.

Paul Minnaar, AM3D, discussed a new crosstalk cancellation system to be played in mobile devices using very closely-spaced loudspeakers. By dividing the source material into two branches and performing crosstalk cancellation at a preferred frequency, he claims that wider image separation can be achieved without losing the timbre of the original stereo material.

Addressing the topic of synthetic audio, Simon Wun, Institute of Infocomm Research, presented an evaluation of iterative matching methods allowing dynamic use for scalable wavetables.

The final session of the conference was a workshop on power-efficient audio chaired by John Oh. During the hour-and-a-half workshop, panelists from diverse backgrounds addressed this important issue from different viewpoints. Kiyoung Choi, Seoul National University, presented issues related to IC design. He introduced an interesting new idea to reduce the number of switches in logic circuits. Jihong Kim, Seoul National University, discussed the software issues of dynamic voltage scaling (DVS) and dynamic power management. Juha Backman, Nokia, asked some fundamental questions about why electronic audio systems have such a low efficiency of around 0.05 to 0.1 percent. He pointed out that the transducer is an important key to higher efficiency. Yonhong Jhung, Tamul Multimedia, presented a low-power implementation of an MP3 decoder using a hard-wired approach. He reported an implementation of MP3 decoding with power consumption as low as 4mW. Pascal Tournier, On-Semiconductor, discussed power savings using Class-D amplifiers. And Pierre-Louis Bossart offered additional input on mobile audio architectures.

The conference ended with closing remarks from AES President Neil Gilchrist. He thanked everyone for attend-
During the short walk from the National Folk Museum of Korea to Kyeongbokgung Palace, attendees pause for a group photo.

Editor's note: The conference proceedings and CD-ROM can be purchased online at www.aes.org/publications/conf.cfm.