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DSP Application Day 2009

e-Conference & Webinar

Milan - September 21, 2009

As has by now become tradition, DSP Application Day aims to inspire students, researchers, teachers, and industrial designers in fields using digital signal-processing technology. The event is a chance for participants to present emerging technologies and innovative applications. A seminar will delve deeper into one specific topic in DSP methods and/or DSP development.

This year's edition will be held online (e-conference and webinar) and will thus give a broad public the chance to learn more about the latest methods and technology for digital signal processing.

Organization:

Mario Malcangi	*malcangi@dico.unimi.it*
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Scientific Commettee:

Goffredo Haus	DICo - Università degli Studi di Milano
Mario Malcangi	DICo - Università degli Studi di Milano
Paolo Rossi	DICo - Università degli Studi di Milano
Francesco Tisato	DISCo - Università degli Studi di Milano Bicocca
Andrea Vitali	STMicroelectronics – Agrate Brianza - MI

Date:	September 21, 2009
Duration:	one day
Location:	on line (http://jli.dico.unimi.it/jli_dspday)
Registration:	Registration is required, free of charge, but on a first-come, first-served basis. To register, send an email message to DSPAppDay2009@dico.unimi.it with “Register DSP Application Day 2009” as the subject.
Web:	http://jli.dico.unimi.it/jli_dspday

e-C O N F E R E N C E

9:00 Opening
Can a tiny embedded microcontroller run a digital signal processing application?

Mario Malcangi
DICo - Dipartimento di Informatica e Comunicazione - Università degli Studi di Milano

Microcontrollers are widespread in embedded applications, where digital signal processing is taking on growing importance. Microcontrollers are an enabling technology because they are inexpensive, consume little power, and fit in tiny devices, although they are not so fast as digital signal processors (DSPs). Softcomputing (artificial neural networks and fuzzy logic) methodologies can help developers design digital signal-processing applications light enough for tiny microcontrollers. A speech-recognition application running on a tiny microcontroller will be illustrated.

9:20 A portable DSP-based device improves music quality for the hearing-impaired

Graziano Bertini, Massimo Magrini, Filippo Paolini
ISTI-CNR, Istituto di Scienza e Tecnologie dell'Informazione, "A. Faedo", Pisa

The Signal&Images Lab of the ISTI-CNR in Pisa has worked for many years in the field of audio-musical signals, including research on analyzing and synthesizing sounds and applications related to multimedia production and performance. One of the topics studied concerns developing methods for improving quality and fidelity when listening to music in certain specific situations.
This article presents our design for an audio device to enhance music listening for the hearing impaired, as compared to commercial hearing aids. It is well known that the performance of modern hearing aids is optimized to compensate for loss within spoken audio range (4 to 5 kHz). This limit was adopted in order to avoid major complications due to miniaturization, high power consumption, and so forth.
Our proposal concerns the design of a portable (not miniaturized) device based on a DSP, which implements adapted gain curves extended through the entire audio band (16 kHz). The apparatus receives incoming music from any "line out," is powered by rechargeable batteries, and produces sufficient audio power to drive normal headphones, thus avoiding the need for hearing-aid earmolds. The device is especially suited to hypoacusic people with mild-to-moderate hearing loss and sufficient residual hearing in higher frequencies, especially for preverbal-age hypoacusic children. This paper discusses our preliminary experiments, describes the prototype's design characteristics, and illustrates several tests.

9:40 Tempo- and time-signature induction in the MP3-compressed domain

Antonello D'Aguanno, Maurizio Botti
LIM – Dipartimento di Informatica e Comunicazione – Università degli Studi di Milano

In this paper we propose a template-matching algorithm to address tempo-induction problems in the MP3 domain. The algorithm is based only on an MP3 window-switching pattern (WSP). The same technique is applied to address the task of extracting time signatures. This problem is also analyzed using a state-of-the-art autocorrelation function in the MIDI domain. All the algorithms presented in this paper work directly on MP3 audio, using only side information. As a result, these algorithms are not time consuming. Experimental results for a range of different musical styles, including rock, jazz, and popular songs, with a variety of BPM tempos and time signatures, are discussed.

10.00 Audio DSP software to compensate for artifacts in digital audio reproduction

Simone Bianchi*, Tommaso Giunti**, Massimo Magrini**
**TangerineTech Engineering ** Associated researchers, ISTI-CNR, Pisa*

This work reports on our studies of acoustic distortion (non-linear frequency response) caused by the listening room in a typical audio-reproduction chain. We used spectral analysis to estimate the environmental response of typical listening rooms to pink noise. As part of the collaboration between ISTI-CNR and TangerineTech Engineering, an environmental equalization procedure using digital high-resolution parametric IIR filters was developed. This research led to audio-DSP software that can compensate the filtering artifacts of most listening rooms. The application was developed for the Apple Macintosh platform.

10.20 Contatto: an interactive music-therapy system for autistic spectrum disorder and psychomotor disability

Massimo Magrini, Leonello Tarabella, Graziano Bertini
ISTI-CNR, Istituto Scienza e Tecnologie dell'Informazione "A. Faedo", Pisa

In the context of Contatto project, commissioned by MousikEssere, the ISTI-CNR Signal and Images Lab developed an active music-therapy system for young patients with autism spectrum disorders (ASD). The system uses a video camera, a FireWire digitalization board, and a Macintosh computer running custom software. During the therapy sessions, the patient moves his body freely inside an empty room. The software uses special algorithms to extrapolate features from the human figure, such us spatial position, arm and leg angles, and so forth. Via a graphic user interface, the medical operator can link these features to sounds synthesized in real time, according to the therapy schema. The system has very low latency thanks to Mac OS X native libraries (CoreImage, CoreAudio). The resulting augmented interaction with the environment could help improve young autistic patients' contact with reality. With a few minor variations and the use of additional sensors, the system could also treat other diseases, such as Alzheimer's.

10:40 DSP Integration of an OFDM communication system

Angelo Consoli, Rui A. Vieira, Christoph Lehmann, Lorenzo Moriggia
SUPSI, Manno (Lugano), Switzerland

The main goal of this work is to build a secure, low-data-rate, radio-communication system, to cover distances of about 10 to 20 kilometers. Its applications will include wide-area monitoring systems and telemedicine. This communication system uses orthogonal frequency-division multiplexing (OFDM), with 16- and 64-QAM modulation. The radio link will preferably use an ISM band in the shortwave range. The system is being developed on a ORSYS evaluation board with the Texas Instruments TMS320 C6713 DSP, expanded with an FPGA, 12-bit AD/DA-converters, and a cryptochip. The OFDM modulation is implemented in software on the DSP, while the encryption is done by the external, on-board cryptochip.

11.00 Voice recognition, recognition grammars, and multimodal development models.

Fabrizio Gramuglio
DotVocal, Genova, Italy

We are used to thinking in terms of two distinct scenarios: the telephone and the computer, equipped with primarily graphic interaction devices. We are currently witnessing the convergence of the two kinds of tool. This trend is accompanied by the rise of a new class of applications/services and of new development models and languages, the multimodal languages that enable us to use voice and video interfaces on a single device.

11:20 Designing VoIP phones with the SPEAr system-on-chip

Stefano Antoniazzi
STMicroelectronics

The new SPEAr™ family of system-on-chip (SoC) by STMicroelectronics provides an innovative solution for designing a wide range of heterogeneous products, avoiding the higher costs of dedicated application-specific integrated circuits (ASICs), while keeping the flexibility of a low-cost customizability for part of the SoC digital logic. This paper describes one specific application enabled by SPEAr, namely high-end VoIP phones for the business market. The application clearly demonstrates the benefits of customizable logic, as well as the suitability of the embedded ARM9E CPU core for digital audio processing, as required to implement the target product. Customizable logic allowed us to integrate hardware functions required by VoIP phones but not offered by common SoC fixed logic. Such hardware functions include digital audio ports and NAND flash interfaces. On the other hand, the ARM9E CPU proved very powerful, suitable for processing audio without embedding dedicated DSP cores.

This paper focuses especially on processing digital audio with software for standard CPU cores that has been equipped with an extended "DSP" instruction set, as offered by SPEAr. The solution allows integrating software optimized for algorithms, such as speech codecs, echo canceling, silence suppression and voice-activity detection. Both narrow-band and wide-band speech processing have been integrated into the final platform. Three-way audio conferencing is also supported.

Finally, an enhanced architecture (currently under study) is described. This solution exploits a dual-core version of SPEAr SoC, where audio processing can be executed in parallel on a second ARM9E core, enabling the main CPU to be totally available for more complex application software.

11:40 Robust, FPGA-based, image-processing algorithm for laser triangulation

Aleš Gorkič, Janez Možina, Janez Diaci
University of Ljubljana, Faculty of Mechanical Engineering,
Aškerčeva 6, 1000 Ljubljana, Slovenia

Rapid laser triangulation is one of the methods for acquiring 3D object surface geometry. It is based on illuminating the object's surface with structured laser light and imaging of the illuminated surface with a camera. The acquired image is usually processed with a PC to obtain a 3D profile. Image processing is a very demanding task for computers because very high bandwidth requires great processing power. We employed in-camera FPGA for image processing, which offers huge performance and power-consumption benefits. This contribution describes a novel image-processing algorithm used for laser triangulation. The image processing is focused on peak detection from image intensity information. The algorithm is divided into five steps: transposing the image to gain 56% more points in a row; FIR filter convolution, which filters and derives pixel intensity; zero crossing to detect pixel position; sub-pixel detection with four-bit precision; and intensity-based points sorting. The algorithm developed is very robust and has only one setting: convolution-filter coefficient set. The selection depends only on laser-line thickness, and the algorithm produces useful results even if not set properly. After processing, bandwidth requirements are cut 150 times, enabling the use of multiple cameras simultaneously.

12:00 DSP implementation of automatic calibration and position calculation for sinusoidal encoders

Ivan Defilippis, Silvano Balemi
SUPSI, Manno (Lugano), Switzerland

Sinusoidal encoders are used to precisely compute the position of linear (or rotating) axes with extremely high accuracy (up to few nm). Sinusoidal encoders must normally be hand tuned and usually suffer from various drifts and non-linearities, which limits their precision. This paper shows how modern DSP microprocessors can be efficiently used for both automatic tuning (calibration) and computing position in a cost-effective way.

W E B I N A R

'Blind' does not mean visually challenged

14:00 Extracting source signals from mixed data

Emanuele Salerno
Istituto di Scienza e Tecnologie dell'Informazione
CNR - Area della Ricerca di Pisa

We introduce the problem of separating a number of individual signals from a set of mixtures thereof, when the mixing coefficients are unknown. Some of the real-world applications where this problem is relevant are then reviewed, along with some of the available estimation strategies. Specific practical needs often require dedicated computing systems.

Extracting the individual signals (the "sources") from a multisensor or multichannel composite signal is not easy, especially when the data are noisy and the mixing operator is unknown. This problem emerges in diverse applications, ranging from audio- and image-processing to telecommunications and biomedical signal analysis.

All the techniques dedicated to solving this problem are referred to as "blind source separation," because the operator that combines the elementary signals is totally or partially unknown. For this reason, assuming a known operator often leads to wrong solutions. Being 'blind' is sometimes better than seeing. However, since knowledge of the operator is unreliable, some assumption must be made about the signals. One of the most popular assumptions is mutual independence, which gives rise to all the statistical techniques based on the independent component analysis principle. The solutions obtained suffer from several indeterminacies, depending on the kind of mixing operator involved and on possible additional available information. Under specific conditions, the independence assumption can also be overridden.

Several practical applications show peculiarities that prevent some classes of algorithms from being applied successfully. Among various difficulties that may be encountered, we mention:
- A poor fit of the signals considered to the assumptions made;
- The presence of particularly strong and nonstationary system noise;
- A noninstantaneous mixing operator, especially if the kernels are totally unknown and channel-dependent;
- A nonstationary mixing operator.