Mistake #3: Digitally controlled analog processing

In 1976 I was asked to design a machine to do digitally controlled analog synthesis. This was a retrofit of a DEC PDP-8E to run a large analog music synthesizer. In hindsight it's obvious this was a really silly idea. If the processing exists in analog or digital and you choose to use digitally controlled analog, you're probably going to regret it twice—once when you build it, because it is difficult, and then again soon afterward, when somebody makes the all-digital solution and puts your machine in the junk heap.

LESSON LEARNED: Design the processor and its programming environment together.

Mistake #4: Creating an overly complex user interface

To allow for manual adjustment of gain and offset, each of the above machine's 24 ADCs and 24 DACs originally had a pair of front-panel potentiometers. They were thrown out of the design for cost reasons and I wrote a software calibration routine instead. The mind boggles at a panel covered with the 96 user-adjustable controls I'd have created without the intervention of the accountants.

LESSON LEARNED: Make sure there's enough processing power for the job at hand, with an ample margin.

Mistake #5: Assuming the processing is fast enough

The above software calibration idea should work fine, but it added 96,000 additions and 96,000 multiplies to the real-time software. On the processor we were using, an addition took 2.6 µs and a multiply took 8.6 µs, so to process one second of data took 1.0752 seconds! Unfortunately software calibration only works if the processor is fast enough. Just pushing a problem into software, or into an FPGA, doesn't guarantee that the resources will be sufficient to solve it.

LESSON LEARNED: Don't mess with hybrid solutions if a digital solution is on the horizon.

↘ VIDEO *http://www.youtube.com/watch?v=LSd4MLVOqFM*

Six world-renowned pioneers of computer and electronic music, including John Chowning, gather at Tulane University in New Orleans to discuss the future of the form—as they saw it in 1967 and as they see it 40 years later

Mistake #6: Designing the hardware before the software

In 1981 I began making a large, experimental ECL processor intended as the engine inside a digital audio mixing console. I planned that the software environment to program this would be graphical and quite advanced for the time.

Unfortunately, however, the software people weren't available when the processor was designed, and several seemingly trivial decisions in the instruction set made the compiler design unnecessarily complex. Every processor I've designed since has been designed together with its programming environment.

When we did the second, TTL version of the processor, which did go into the product, we designed the processor and compiler together, to much greater success. We also made a simulator so we could "run" code on the design before we made any hardware.

LESSON LEARNED: Keep the user interface simple.

But despite those lessons learned, I still make mistakes—and I still learn from them—every day. ■

Peter Eastty has been working in professional digital audio since the '60s. Stints with Electronic Music Studios, IRCAM, Systems Concepts, Solid State Logic and Sony Pro-Audio have led to his position as CTO of Oxford Digital. Among his interests are the design and programming of fast signal processing, numerical algorithms and testing the rev limiter on his Mazda RX8.

PC AUDIO SUBSYSTEMS

Evolution and design

By Rob Maher

TWENTY YEARS AGO, when someone wanted real-time audio on a personal computer, it typically meant toggling bit 1 in I/O port 97 to generate a pulse waveform through the PC's small built-in annunciator speaker. Ten years ago, real-time audio on a PC typically meant buying a PCI-bus plug-in sound card. In the past decade, however, the vast majority of PCs have emerged from the shipping box ready to play with motherboard audio solutions based on the Audio Codec (AC) '97 specification and its successor, the Intel High Definition Audio (HDA) multichannel spec.

PC audio: The transition to AC'97

When introduced by Intel in 1997, the AC'97 specification stunned the PC audio business by encouraging motherboard manufacturers to place relatively inexpensive mixed-signal "codec" chips right on the motherboard itself. The digital controller portion of the AC'97 spec resided right in the "southbridge" silicon of the I/O Controller Hub (ICH), while the analog-to-digital and digital-to-analog functionality needed for audio and modem features was placed in the audio codec chip. AC'97 was geared toward supporting monophonic microphone input and two-channel (stereo) audio I/O features, 16- to 20-bit resolution, and 48-kHz sampling rate per channel.

Motherboard audio provided the big advantage of ubiquity: essentially any PC could have low-cost audio I/O with standardized hardware features and software device drivers. Operating system software could also be standardized to use the audio codec as an easy plug-and-play device.

However, not all end users were

completely satisfied. Audio quality was not always up to the standards of hi-fi enthusiasts and serious gamers, because the PC motherboard is an inherently hostile electrical environment for low-noise audio. AC'97 systems could also exhibit annoying audio glitches, clicks and dropouts when the host processor was heavily loaded—during a graphics-intensive game or screen update, for instance. That's not good when you're trying to enjoy a DVD with Dolby Digital 5.1 sound.

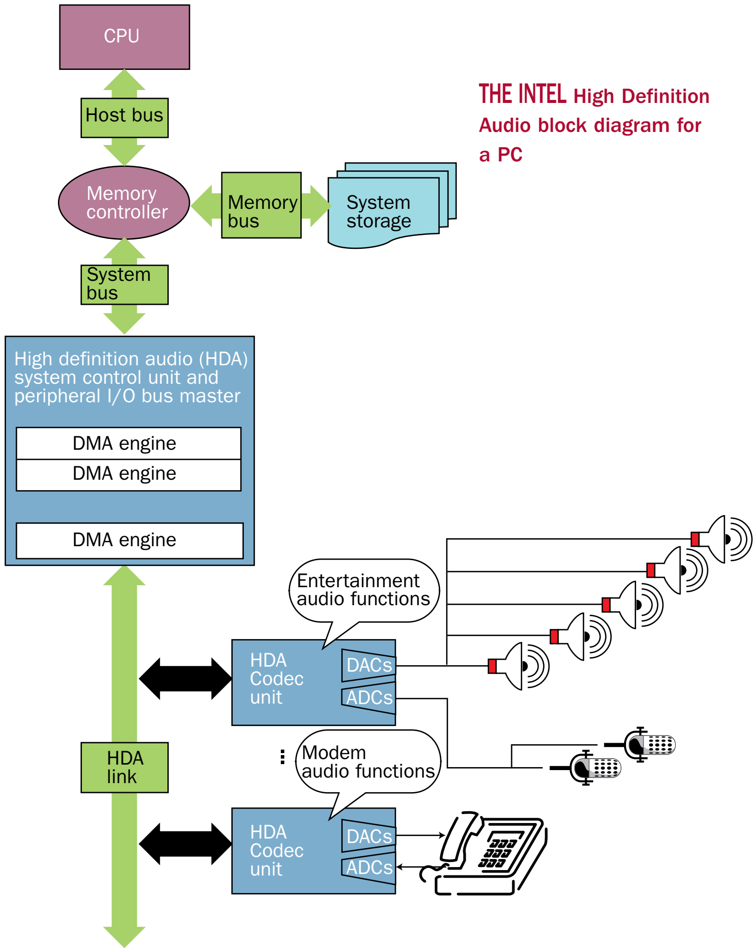
What's new? Multichannel audio, of course

Since 2004, Intel has promulgated the High Definition Audio (HDA) standard (see figure) to address expanding end-user expectations. Intel HDA shares the AC'97 heritage of motherboard sound solutions, but adds higher-performance flair for home entertainment support. The HDA spec allows for eight channels of 192-kHz/32-bit audio output.

The HDA link is a five-wire digital serial interface between the HDA controller, typically part of the ICH core logic chip, and the HDA codec chip. The HDA link employs a controller synchronous bit stream locked to a 24-MHz bit clock (BCLK) provided by the HDA controller. All of the audio input and output data streams are isochronous with a defined 48-kHz frame rate SYNC signal.

Typical HDA systems deployed in PCs provide five to eight analog audio channels (7.1 support) onboard, with selectable 48- to 192-kHz/24-bit analog audio outputs. Some HDA codec chips and motherboards provide S/PDIF digital audio input/output via coaxial (copper) connectors, optical connectors or both. What's more, the design of the hardware and software layers of the audio subsystem has greater sophistication, lowering the likelihood of performance-related glitches and dropouts.

Hardware design demands impeccable analog engineering to maintain signal integrity in the harsh PC environment



THE INTEL High Definition Audio block diagram for a PC

Besides raw multichannel audio support, contemporary PC audio systems must also provide multiprogram features for concurrent playback. For example, a home entertainment computer might be used to play multichannel surround sound for a DVD movie in the family room while a user in the kitchen or study enjoys the sound from an Internet video-sharing Web site. Similarly, serious online gamers will choose to experience the game's 5.1 audio soundtrack over loudspeakers while the live Internet audio chats flow separately—but simultaneously—through the user's lightweight headset.

PC audio subsystem design challenges

Clearly, the challenge for audio subsystem designers and programmers is to ensure that the multiple concurrent audio channels get where they need to go—and to make it easy to set up, debug and maintain. Hardware designers must practice impeccable analog engineering to maintain signal integrity in the harsh PC environment.

Beyond selecting the audio codec chip, the designer must also choose the suitable routing, location and style of the analog audio connectors

Rob Maher, Ph.D., is head of the Montana State University Department of Electrical and Computer Engineering, where he teaches and conducts research in audio DSP. He was formerly engineering manager for audio products with 3Com-U.S. Robotics and vice president of Engineering at EuPhonics.

↘ **VIDEO** *http://www.intel.com/design/chipsets/audio/nav.swf*

This animation from Intel demonstrates how HD Audio provides surround sound and advanced audio functions for the home and office environments

and digital I/O if S/PDIF is required.

Codec manufacturer layout recommendations generally suggest physically separating the digital interface portion of the codec chip—such as the HDA link and S/PDIF connections—and the sensitive analog audio input/output pins in order to minimize inductive crosstalk of the high-frequency digital signals into the analog circuitry.

Similarly, choosing the proper input and output audio DC-blocking capacitors can have a substantial positive effect on low-frequency performance. Most codec manufacturers specify using MLCC caps with X5R dielectric (very low-effective series resistance and inductance) on the audio input side, and relatively high-capacitance electrolytic devices (>10μF) on the outputs. With care, the hardware designer can truly deliver on the promise of high-definition audio in the PC environment. ■

MPEG SURROUND

A new standard for surround sound

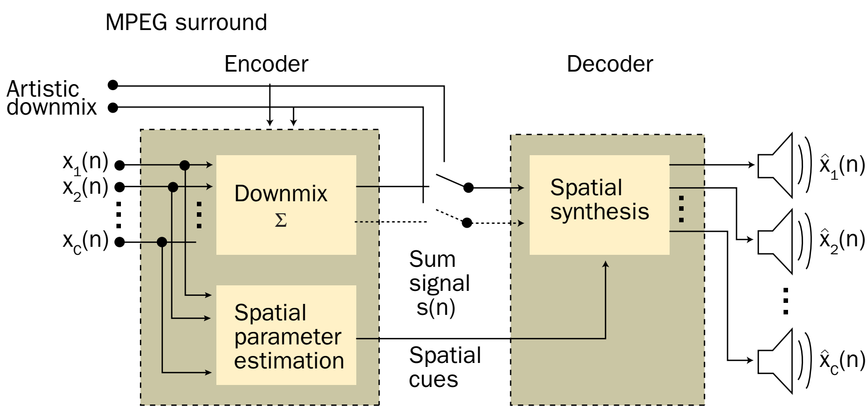
By Matthias Rose

UNTIL RECENTLY, the implementation of surround sound for radio broadcast has faced considerable challenges. Compatibility issues, poor surround quality and the high bit rates required for surround transmission, among other factors, have conspired to put the brakes on the multichannel juggernaut, leaving analysts and professionals to wonder exactly when 5.1 radio might finally become a practical proposition.

That moment has now arrived, thanks to a new MPEG technology substantially developed by Dolby Laboratories, Fraunhofer IIS, LSI and Philips.

MPEG Surround arrives

The result of several years of painstaking work, MPEG Surround (*www.mpegsurround.com*) has been developed specifically to enable high-quality surround sound at stereo bit rates, designed for a straightforward implementation with existing stereo or mono infrastructures. The fully backward-compatible technol-



WORKING PRINCIPLE OF MPEG SURROUND: On the encoder side a mono or stereo downmix is created or, alternatively, an externally created downmix signal may be used. Spatial parameters describing the sound stage are estimated. The downmix is then transmitted together with the side information. On the decoder side, the spatial parameters are used to recreate the original sound source. Legacy receivers ignore the parameters and play back a stereo signal

Matthias Rose heads the Marketing Communications group of the Audio and Multimedia division of Fraunhofer IIS, a Germany-based research institute credited with the development of the MP3 coding algorithms and co-development of AAC (*Advanced Audio Coding*) as well as MPEG Surround technology standards.