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Digital Audio Processing II
E85.2601, Prof. K. Peacock

"The World Beyond 20kHz"

by David Blackmer

I've known for awhile now that the currently standard sampling frequency of 44.1kHz puts limits on the sonic fidelity of recordings. Blackmer's article, though, made me realize for the first time the exact motivations behind the 96kHz+ argument. My shortcoming was that I had always assumed that the push towards a higher sampling frequency was for something in the frequency domain; this article, however, made me realize that the need for a higher sampling response arises from the time domain (phase linearity, for example). More specifically, it is the slow impulse response (the time it takes to respond to an input) of many audio systems today that limit their fidelity.

I guess the main way that this article ties into our digital audio process seminar is superficially that it argues for a higher sampling rate. This much is obvious. The deeper point brought up by this article, however, concerns the unending search for audio perfection. Perhaps it may seem defeatist to say so, but it seems that once a technological standard has been established in the music industry, it is not long before consumers, engineers, and artists are clamouring for more; whether this more is in the form of more surround speakers, higher sampling rates, or increased user flexibility, the theme of continued improvements and accomplishments remains the same. This notion of constant development is actually the focus of my article abstract this week, entitled "The Future of Audio."

In conclusion, I was glad I got a chance to read this article. It cleared up a lot of questions about the nature of human hearing. I also thought it was oddly ironic that it is the phase accuracy of low frequencies which plays an important role in the higher sampling frequency since I was always under the impression that the higher rates were related to ultra-high tones. I guess the only thing left to do is buy a pair of those Earthworks microphones!

Article Summary:

"Audio in the year 2000" by George Petersen
Mix Magazine Supplement, January 1999, pg. 50-82.

In this article, George Petersen speaks to seven technologists from divergent fields of audio and questions them about many topics concerning the future of audio. The responses range from practical, foreseeable predictions to uber-avant-garde statements (like accurate soundfield reproduction through the use of thousands of small, flat speakers mounted like wall-paper). The main thrust of this article, however, is the certainty of change. The music industry is currently in a major state of flux and far from being focused. Apparently, nothing on the horizon truly seems to offer a standard relief from the emerging new products and digital processes.

Concerning the 192kHz phenomenon, most experts predict that within the next two years, all professional products will be operating at least on elevated sampling rates such as 96kHz. This need arises not from the need to capture higher frequencies (since they truly are inaudible), but the desire to more accurately represent timing information. As humans can perceive a difference of five microseconds between the two ears, this number translates to a distance shorter than the distance between 96kHz.

Speaking of the increase in digitization of professional audio applications, mention is made with reference to live sound. Digital public address and acoustic systems are still in the infancy of their design but promise a revolution to the operation of live sound. The key to this revolution is the development of networking. Yamaha, for example, is currently working on its mLan spec, which will provide "the capability to send sample-accurate AES digital audio, MIDI, and other control information over IEEE 1394 (Firewire)." Other networking possibilities exist. Gigabit ethernet provides the opportunity, for example, to put thousands of channels of audio around a theme park. Also, companies' audio systems in conference halls can be tied together.

The final digital audio subject addressed in this article concerns DSP Integration. Most currently available digital filters are only models of analog predecessors--the IIR model (Infinite Impulse Response). Digital technology, however, has the potential (requiring a great amount of processing power) to implement FIR models (Finitie Impulse Response). With FIR, there is no phase shift so that one could thus have infinite slope crossovers with no phase distortion. On the other hand, complaint is made of the increase of functionality of digital processing with sacrifice to usability. Many people complain that DSP too often reveals too much of the "guts" of the processing system, providing too many choices (!) for operators and thus sacrificing the ease of achieving a simple good effect from the devices.

As far as my informed skeptic view is concerned, I feel much of what was discussed in this article was interesting, but too far in theory. Some of the predictions almost seemed more like science fiction than upcoming new products. Considering that these statements were coming from respectable company representatives, I was surprised that most did not take a more conservative stance about the industry. Digital audio really seems to have put a genuine panic in companies afraid of what might be invented before they can patent it first. Even if all the futuristic technology that they portend is introduced within the next ten years, it will assuredly be a bit buggy, slightly unmanageable, and scary to most consumers. So bring it on.