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Digital Audio Processing
E85.2600, Prof. K. Peacock

Discussion of Ricardo Santoboni Lecture

I must admit to having enjoyed Santoboni's lecture two weeks ago. My enjoyment of his talk, though, was more a result of his musical discussions than his technical explanations. The concept of using something other than a chord to define tonic versus dominant relationships was particularly fascinating. In his case, "connotated" sounds (those already familiar to the listener) were defined as representing the tonic or consonant harmonic area. Conversely, "neutral" sounds (those foreign to the listener) were defined as representing dominant or dissonant areas. An odd byproduct of such a system is that neutral sounds, upon being presented many times in a piece of music, shift towards becoming connotated sounds. Dissonance, therefore, eventually sounds like consonance upon multiple presentations. While this theory is interesting enough to stand alone, I had trouble distinguishing it from any other post-tonal theory once it was applied to a real musical composition. In other words, Santoboni's music did not sound much different to my ears than most other music from the latter part of this century despite his highly developed systemization of sounds. Perhaps my ears have been overly tonicized by years of classical and pop music. On the other hand, perhaps Santoboni should have tried to implement his post-tonal ideas to a simpler harmonic structure. Form is a very important part of music, and it was the form of his music that I had trouble following. I would be interested to see the results of mapping his consonant/dissonant versus connotated/neutral theories against a minuet by Mozart or a Strauss waltz. With such obvious formal constraints, I think it would be easier to judge the value of his theories as compared to the tonal music theory that has been in place for at least the last four centuries. As far as his technical discussion of the M.A.R.S/A.R.E.S software was concerned, I had trouble following him. Perhaps an overhead projector that would have enlarged the computer screen would have helped. In general, I thought he very carefully explained his compositional practices but just kind of jumped into the software, leaving me a bit confused.

Synopsis and Implications of "Firewater Update" by Michael Clagett

The existence of a tangled mess of wires plagues every recording studio and many homes in the world. Until someone invents some sore of information transmission method which surpasses the use of point-to-point wiring, humanity will have to live with wires to connect various audio and video devices (including MIDI). It is the main goal of the Firewire project to develop a standard that will allow any and all audio and video data to be transmitted along a single cable, thus organizing the interfacing of different equipment and generally cleaning up the area behind such equipment (which is now a mess of wiring). The larger need for such a system, of course, exists in the professional field where multiple audio/video systems are continually being linked together. Since the professional audio industry, however, has a much smaller purchasing power than the general public at large, the current Firewire standard is being developed to consumer specifications. While the development of a consumer wiring standard does not preclude the development of a professional standard at a later date, it is somewhat discouraging to see effort being spent on a less pressing need.

The basic Firewire specification limits professional usage on a number of levels. Foremost, the standard only allows a 4.5 meter cable length. This short distance doesn't cater to the long cable length needs of the average recording studio where total distances often reach over a hundred meters. The second problem with the current Firewire spec is that it only clocks data 8,000 times a second. While having a clocking speed of only one-sixth (48,000 divided by 8,000) the audio sampling rate may cause a forgivable amount of jitter in a consumer system, the professional audio community will not tolerate such a rate.

Hopefully these impediments to a professional specification will soon be overcome. The implications of a universal wiring standard are indeed far reaching. With so much of the music recording process being increasingly dominated by digital methods, computers and computer technology are factoring into audio more prominently. Since the computer, video, and audio industries previously developed separate means of information transmission, it becomes important to standardize and unify these transmissions. With such a standarization, moving between the audio, video, and computer realms will be easier, more efficient, and thus less time consuming. Without the need to worry about interfacing equipment, Firewire promises to allow the audio professional to concentrate more on the truely creative side of audio applications and less on the mundane task hooking up wires.

Explanation of Figure 7.19 in "Art of Digital Audio" by John Watkinson

The figure on page 369 in Watkinson's book shows a detailed list of 48 subframes. A subframe includes one audio sample and all related information such as subcode, channel number, bit rate, etc. Two subframes equal one frame, and therefore this list shows 24 frames. A subframe for a Compact Disc audio sample begins with a synchronizing pattern that alternates between the two audio channels (A and B). Following this intial sync is the auxillary data and 16-bits of the audio samples themselves. For higher bit applications, the byte of aux data can be used to encode 20- or 24-bit samples. Commonly, however, this aux data is used to provide talkback for broadcasters (although in the CD format, this application is unnecessary). The audio samples are arranged least significant bit first, proceeding to most significant bit last. This ordering of the samples facilitates mathematical operations (such as is used in mixing two samples) since carry-forward adding functions can be delayed until the next higher bit. Following the audio data is a validity flag bit that currently is not used for much except identifying non-analog compatible data (such as exists on the CD-I format). The user bits come next, where the subcode is transmitted. Subcode for CDs assists in locating the beginning of different music tracks on the disc and providing a catalog of their duration and location on the disc. The penultimate part of the subframe are the channel bits. This information transmits the sampling rate of the audio and the clock accuracy. Finally, a parity bit finishes the subframe data. The parity bit produces even parity over the subframe by making an even total number of ones in the subframe. If the total number of ones were even up until the parity bit, the parity bit becomes zero. Vice versa, if the total number of ones not including the parity bit were odd, the parity bit becomes one. The main purpose of the parity bit is to make successive sync patterns have the same polarity.