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Digital Audio Processing I
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Research Status Report #2

In my first research report, I proposed to study and explore how manufacturers of digital technology are incorporating the inherent aesthetic advantages of analog tape recording into the design of their products. This topic of choice seems to be a premature investigation on my part, though, because I am relatively unfamiliar with the mechanics and advantages of digital technology itself. As my ultimate goal is to compare analog tape recorders to digital tape recorders, I embarked this week on a short foray into digital tape recorders.

To understand how a digital tape recorder works, one needs to fully understand the theory behind digital encoding of sound. From reading the introductory chapters to digital tape recorder compendiums, I found that the Watkinson text, <u>Art of Digital Audio</u>, had yet to reveal to me important facets of digital operation. A basic concept: the Nyquist theory states that in order to receive an accurate sampling of a particular frequency, the sampling rate has to be at least twice this frequency. For example, since human hearing can perceive frequencies only as high as 20kHz, one needs to set the sampling rate for at least 40kHz.

Another basic concept to PCM is the problem of aliasing where frequencies higher than the Nyquist limit are folded down into the audible range. These shadow frequencies can be calculated by subtracting the Nyquist limit from the frequency undergoing aliasing. For example, at a sampling rate of 40kHz, a 30kHz tone is folded down to 10kHz (30kHz-20kHz). This folding effect is produced by the higher soundwaves fitting somewhere in between the pulses of PCM.

To combat this aliasing, digital systems first introduce to the signal flow an antialiasing filter, which basically filters out frequencies which would cause aliasing. Since these frequencies are always ones that are necessarily higher than the Nyquist limit, this antialiasing filter basically acts as a low-pass filter centered around the Nyquist limit of the system. Problems with low-pass filters, since they are analog devices in the chain (pre-ADC), include 1). low-pass ripple: slight random increases to amplitudes for the frequencies which are passed; 2). stopband ripple: the act of letting pass still some of the unwanted aliasing frequencies; and 3). rolloff: the characteristic of affecting to various degrees a distinct bandwidth of frequencies instead of just one frequency. In order to combat a problem such as rolloff, samplers can be set to slightly higher sampling rates, as with 44.1kHz, in order to shift this rolloff area outside of the audible spectrum.[1]

Another simple concept related to digital audio is quantization noise. To clarify, there are two types of noise in any audio system: idle and dynamic. Idle noise is present when no input signal exists; it is the background noise generated by the system at rest. Dynamic noise, on the other hand, is the noise generated by specific inputs, usually those which exceed a certain threshold of capability by a recorder or output device.[2] A sort of reverse dynamic noise occurs due to quantization. Quantization necessarily adds noise to any signal due to its effect of approximation. For example, if a voltage level of 0.25 volts is quantized to 0.3 volts, then a noise level of .05 volts (or 20% of the original signal) is introduced. Since a quantization level is usually fixed, it acts irrespective of the amplitude of the given signal; thus, a higher amplitude signal will be divided into more quantizations than a lower level signal. "The error floor of a digital audio system differs from the noise floor of an analog system, because in a digital system the error is a function of the signal."[3] To combat the noise effects on lower amplitude signals due to fixed quantization levels, the concept of "floating point converters" was developed. With this method, the amplitude of the incoming signal is monitored and the quantization level adjusted accordingly to allow for the maximum degree of resolution.[4] The main problem with floating point converters, however, is that the information required to program and code such devices has been historically higher than is required to simply add a few more bits to the quantization level and thus solve most resolution problems.

There are many other facets to digital audio with which I feel I need to become more familiar before I begin a true study into its relation to analog processes. The use of oversampling, for example, is widespread in many digital recording machines and serves as an alternative to high wordlengths during quantization.[5] I believe, therefore, that further study into "authoritative bibles" other than Watkinson's Art of Digital Audio will be necessary to truly elucidate for me the workings of this medium.

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[1] Schetina, Eric S. The Complete Guide to Digital Audio Tape Recorders. Prentice Hall: Englewood Cliffs, NJ. 1993

[2] Strawn, John, Ed. Digital Audio Engineering. William Kaufmann, Inc.: Los Altos, CA. 1985

[3] Pohlmann, Ken C. Principles of Digital Audio. McGraw-Hill, Inc.: New York, NY. 1995

[4] Blesser, B. A. "Digitization of Audio." Journal of the Audio Engineering Society. vol. 26, no. 10: pp. 739-771. Oct. 1978

[5] Rumsey, Francis. Tapeless Sound Recording. Focal Press: London. 1990