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Chapter 4 Abstract

One of the fundamental concepts of this chapter is data reduction. Ideally, the reduction of data is obviously something to be avoided. Data reduction always runs the danger of also reducing the quality of sound inherent in the system. Unfortunately, size and economics often dictate that the amount of information on a compact disc, for example, cannot all be contained on a MiniDisc. Compression and data reduction is therefore necessary to fit the same information from the medium with higher storage capabilities to the one with lower storage capabilities. The true art of data reduction involves both deciding which information is the least important to ensure accurate sonic reproduction and developing ways of coding this information in increasingly efficient ways. A keen understanding of psychoacoustics is necessary for the art of data reduction because it is the perceptibility of human hearing that is the defining factor of sonic performance. Therefore, data reduction often uses the masking qualities of the ear (which fails to register energy in a sound band when more energy is present in a neighboring band) to determine and delete the redundancy inherent in PCM systems. What good is it of encoding sounds if the ear will not hear them anyway? Some audiophiles may balk at the idea of selectively coding a waveform, but just remember that data reduction already takes place on a compact disc which uses only 16 bits per sample and ignores frequencies above 22.05 kHz. The methods for data reduction are numerous, and I will touch upon a few of them here.

One method of reduction is termed non-uniform coding. Through this practice, lower level signals are left untouched while higher amplitude signals are increasingly attenuated. Since the signal to noise ratio of any system is calculated using a full signal, lower level signals usually have increased amounts of perceptible noise while higher level signals have a smaller percentage of noise that is masked by the higher amplitude signal. By reducing only the higher gain levels, a more constant noise level can be obtained without sacrificing the transparent quality of the system.

Another method of data reduction is called floating-point coding. With floating-point, two approaches can be taken: block and non-block. The basic premise of floating-point conversion is that the binary number is converted into an exponent and its mantissa. The more complicated type of floating-point conversion is the non-block system. Through non-block FPC, each sample is computed to find the correct exponential level. With block FPC, a block of samples is taken to have the same exponent. The danger of block FPC is obviously that some samples may lie outside of the exponential level which was chosen to code that specific block of samples. If this situation occurs, distortion will result through signal clipping. One of the advantages of block FPC, however, is that it takes far less hardware to implement and can be more easily implemented in real time scenarios since the delay introduced by the calculation time of a proper exponential level is reduced.

With predictive coding, the use of sigma-delta modulators is employed. In other words, the differences between samples is recorded instead of the level of the whole sample. If a high amplitude signal doesn't change from one sample to the next, only a small wordlength is required in sigma-delta to convey this lack of change whereas in regular PCM coding, a long wordlength would be necessary to capture the complete signal from one sample to the next. The predictor works by using the last sample as its prediction for the next. The difference between the last sample, or prediction, and the actual following signal is then recorded. Basically, the predictor is finding the redundancy in a system. If a sine wave of an instrument changes quite slowly, then the sampling frequency does not have to be so high. Similarly, this slowly changing sound will be only minimally coded by the predictor since it is only coding changes in sound, not the full sound itself. An even further refinement of the predictor process is to alter the quantizing steps to match the level differences. Thus, will a small change in amplitude, the predictor can use smaller quantization levels for increased resolution. Conversely, larger steps are necessarily used for larger changes in amplitude.

Other methods of coding, such as sub-band and transform are also used in data reduction techniques. The explanation of the mechanics of these and other methods, as well as even finer points on digital audio, will have to be saved for a later date.