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Digital Audio Processing
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Abstract:
Chapter I, <u>The Art of Digital Audio,</u> by John Watkinson

This first chapter serves mainly as an introduction to the benefits and basic concepts behind digital audio. Before understanding these benefits, though, the limitations of analog audio must be first addressed. Analog's main fault lies in its susceptibility towards degradation. Since analog information is carried through infinitely varying mediums (such as changes in voltage or magnetism), no distinction is possible between the original input and any slight variations (or noise). Digital information, on the other hand, has only two states: on and off. This "on and off" state can be represented through the binary system (base 2) in which each digit (or bit) can have only a value of 1 or 0. Without allowing any intermediate states, noise in a digital system below 50% of the source signal can be easily rejected by comparing the information to a threshold. This threshold accepts only "on" or "off" state information and can thus reform any minorly corrupted signal due to jitter. The most common method of converting sound (which exists in the analog domain) to binary information is through pulse-code modulation (PCM). Through PCM, the audio signal is measured at a fixed rate (sampling frequency) and stored in digital words of fixed bit length. Because this sampling rate is always constant, analog flaws such as wow and flutter can also be completely eliminated from a recording system by merely reclocking information to this average sampling rate.

Every digital audio system relies on its analog-to-digital and digital-to-analog converters (ADC/DAC). Between the ADC and the DAC, a digital system reorders the bits of a sample into a continuos stream of information. In order to cause this transformation, the system itself must be running at a frequency multiple of a least the sample wordlength. Therefore, in a 16 bit 44.1 kHz sampling system, the internal information must be transported digitally at a minimum rate of 16 X 44.1 kHz, i.e. 705.6 kHz. A common method of storing this information is in random access memory (RAM). The memory requirements on RAM for music are quite prodigious, considering a megabyte contains only about 1.05 million bits while high quality music has an information rate of at least 705.6 Kbits/second. An alternative to using RAM as storage would be in the form of a programmable delay. Through this method, information is written into a RAM address and read from the RAM at a later address. These different address points cause a specific amount of delay which can be useful in various audio applications. To create a continuous delay, address points return to zero after the RAM becomes full.

Other methods of digital storage rely on clock based techniques. By the time compression method, two RAMs are used to control the flow of information. Initially, samples are read into one RAM at the sampling frequency. Once this RAM is full, samples are read into the other RAM at the same rate. While the second RAM is being filled, the first can be read at a necessarily higher rate than the second is being written; this higher read than write rate creates time compression. Again, for 16 bit 44.1 kHz sampling, the read rate must be higher than 705.6 Kbits/sec. Since the information from the first RAM is completely read before the second RAM has finished being written, a delay is introduced between the first and second RAM's information. This delay introduced by time compression can be used to add coding information to a recording, allow a rotary-head recorder to switch heads, correct errors in the samples, etc. As compared to time compression, real time transfer between digital audio machines is sometimes necessary, such as using a digital mixer with a digital playback machine. In order to facilitate an accurate link between two digital devices, synchronization through common sampling rates is needed. A basic solution are exemplified in "external reference locking processes" which control tape speed based on read address versus write address differences.

Because digital audio shares the same imperfect mediums as analog recording, digital audio is subject to the same dropouts and random large amounts of noise as analog. The method for correcting digital errors is trivial since binary information only has two states, i.e. if one state is wrong the other must be right. With digital audio, the challenge is therefore the process of correctly identifying which bits are in error and developing methods to reduce this error. Pure error correction is a function of redundancy, basically producing inaudibly corrected samples. When error correction is not possible, concealment is used. Concealment depends on interpreting the missing sample from the samples which surround it. Since audio waveforms change is predictable and limited ways, these interpreted samples can be determined with a high degree of probability. In order to reduce the risk of two adjacent samples succumbing to error (and thus hindering the concealment process), samples are often divided among multiple groups. One method shuffles odd and even samples to different tracks, allowing for recreation of every other sample if a whole track is lost to error. Due to the nature of high-density recorders having larger sized dropouts, more complicated methods of sample separation are sometimes necessary. For example, interleaving arranges incoming samples into rows which can be stored in columns; the order is reversed for outgoing information. Such sophisticated techniques of data isolation necessarily cause a time delay in digital machines shifting confidence replay up to one-tenth of a second from the input in some recorders.

Digital audio also presents new and unique intricacies of data storage, mostly engineered to compensate for lack of memory space for audio information. Channel coding deals with representing parallel tracks in the more convenient form of serial information. In channel coding, silence is given a particular waveform so as to correspond with other relayed information. Data reduction, on the other hand, uses a more straightforward method for saving memory by rejecting the three-fourths to four-fifths of PCM data that would be normally rejected by the human ear due to psychoacoustic effects. Digital compact cassettes and mini-discs are examples of consumer recorders using data reduction techniques to reduce the amount of information storage necessary and thus the size of the digital machine. Digital Audio Broadcasting (DAB) also benefits from data reduction since the data rate of PCM audio is too high for the transmission bandwidth of DAB.

As digital bits are so easily represented by universal on or off states, many mediums exist upon which to store this information. Before the advent of high-density digital recording, the PCM adapter was an early solution to the needs of digital audio's wide bandwidth. The PCM adapter made use of the FM modulator on video recorders to represent ons and offs with black and white levels. Hard disk is a more convenient method of binary storage, mostly for its effectiveness in non-linear editing. Since the instantaneous rate of a disk drive far exceeds the converter's sampling rate, access to any part of the audio is much quicker than almost all other mediums. Digital tape recorders such as DAT recorders provide a more efficient storage medium than hard disk yet sacrifice the ease of editing capability. Open reel digital tape recorders were more easily edited than rotary-head digital recorders but were overly bulky while still falling far short of the hard disk's editing capabilities. In general, however, the potential of digital audio portends a compact and inexpensive future for all audio mediums.