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Chapter 11 "Art of Digital Audio" Abstract

John Watkinson's eleventh chapter was a rather brief look into the methods and processes of editing digital audio. The basic nature of editing derives from the early tape cutting procedures of analog tape. Punching in on a mulit-track recorder to correct mistakes, mixing dialogue with sound effects, and the assembly of master recordings are all examples of common requirements made on editing devices. Basically, audio editing is aligning the proper sounds at the proper times. Digital editing meets this demand in a necessarily much more complicated way than its analog counterpart. The result, however, is a far more user flexible system that truly empowers the editing engineer.

The most convenient digital editing method is performed on random access media such as hard disk recorders. With these machines, the recorded data is copied onto various pages of RAM where it can be altered an modified without concern for adversely affecting the original audio data. When the edit has been defined, the editor crossfades between samples to provide a smooth musical transition. This technique is often referred to as non-linear editing. The term non-linear refers not to the nature of the editing process but to the non-linear nature of the storage medium.

Editing on recording media (such as open reel-to-reel recorders or RDAT machines) involves at more complicated process since error correction techniques such as interleaving and codewords have been written to the storage medium. The block-based nature of error correction, however, does not easily allow for real-time access to the musical sequence. Also, editing on such media cannot be performed directly on the media itself. The tape from an ADAT machine, for example, cannot merely be spliced in the method of analog tape editing. The audio samples must be de-interleaved, corrected as per the needs of the edit, and then re-interleaved and written back to the medium. This process is know as read-modify-write. Sometimes, when audio is contained in large data blocks, even the smallest edit requires reading and rewriting the entire audio block.

The operation of editing with digital audio must simulate the tape shuttle methods of analog recording in order that engineers can precisely locate edit points. Samples are continuously read into the editor's memory that is addressed by a counter which repeatedly overflows. Since engineers typically need to hear about thirty seconds of audio to correctly pinpoint an edit point, the storage requirements of stereo PCM audio in the editing system approach 5 MB. Unfortunately, this requirement represents a significant cost to editing hardware. To reduce the size of memory needed, edits usually deal with mono signals that have been converted down to an 11 kHz sampling rate. As with all downward sample rate conversions, an anti-aliasing filter needs to be added to the chain. To simulate the analog tape jog editing method, a scrub wheel (or rotor) is used. The memory addresses in the editor will change at a proportional rate to the speed with which this scrub wheel is turned. Sound can thus be heard forwards and backwards as necessary.

RDAT editing presents its own specific brand of editing problems to overcome. Editing can only be done at the beginning of interleave blocks (commonly known as a frame) which contains two diagonal tracks. Professional machines use two sets of drum heads to address this situation. In normal recording, the first set of heads records while the second set serves to provide confidence playback of the program material. In editing, the situation is reversed; the first set of heads reads the material (de-interleaving, etc.) while the second set of heads writes the new PCM data. Crossfades between the old and new audio data to smooth out the sample stream.