AE16 DIGITAL AUDIO WORKSTATIONS

1. Storage Requirements

In a conventional linear PCM system without data compression the data rate (bits/sec) from one channel of digital audio will depend on the sampling rate and the resolution. E.g. a system operating at 48kHz using 16 bits will have a data rate of:

48,000 x 16 = 768 Kbit/sec or 5.5Mbs per min

2. Usage of Storage capacity on disc versus tape

A 30-minute reel of 24-track tape using 48kHz/16 Bit uses 396MBs only when all 24 tracks are recorded for the full 30 min. With a disk drive the total average storage capacity can be distributed in any way between channels. In fact one cannot talk of 'tracks' in random access systems since there is simply one central reservoir of storage serving a number of channel outputs when they are required.

In disk-based systems, storage capacity tends to be purchased in units of so many megabytes at a time, corresponding to the number of single channel minutes at a standard resolution. The system may allow an hour of storage time divided between 4 channel outputs, but the total amount of 'programme time' to which this would correspond depends on the amount of time that each channel output is being fed with data.

Disk Storage

All computer mass storage devices are block structured, i.e. they involve the dividing up of the storage space into blocks of fixed size, (512 – 1024 bytes) each of which may be separately addressed. In this way information is accessed quickly by reference to a directory of the disks contents containing the locations of blocks of data relating to particular files.

3. Digital Audio Requirements

Samples, since they are time-discreet may be processed and stored either contiguously or non-contiguously, provided they are reassembled into their original order (or some other specified order) before analogue conversion. Thus digital audio is ideal for storage on block-structured such as hard disks, provided that buffer is employed at the inputs and the outputs to smooth the transfer of data to and from the disk.

4. Audio Editing

Audio Editing may be accomplished in the digital domain by the joining of one recording to another in RAM, using the buffer to provide a continuous output. A fast access time disk drive (under 20ms seek time with no thermal calibration) makes it possible to locate sound flies at different times and play them back as one continuous stream of audio without any audible glitches.

5. Multichannel Recording

Multichannel Recording may be accomplished by dividing the storage capacity between the channels. During system operation, audio data is transferred to and from the disc via RAM using the Direct Memory Access (DMA) controller and buss, bypassing the CPU. Data is also transferred between RAM and the audio interfaces via the buffers under CPU and user command control. During editing, fading and mixing data is written from the disc to the DSP unit via RAM, which in turn passes it to the buffered audio outputs.

6. Concepts in Hard Disk Recording

The Sound File

This is an individual sound recording of any length. The disk is a storage in which no one part has any specific time relationship to any other part - i.e. disk recording does not start at one place and finish at another.

Sound File Storage - The Directory

A directory is used as an index to the store, containing entries specifying what has been stored, the size of each file and its location. Within the directory, or its sub-indexes, the locations of all the pieces of the sound file will he registered. When that particular file is requested the system will reassemble the pieces by retrieving them in a sequence.

Buffering

Buffering ensures that time-continuous data may be broken up and made continuous again. A buffer is a short-term RAM store, which holds only a portion of audio data at any one time. Buffering is used to accomplish the following tasks:

- Writing/Reading - Disk media require that audio is split into blocks (typically 512- 1024 bytes) and this is achieved by filling a RAM buffer with a continuous audio input and then reading out of RAM with bursts of disk blocks which are written to disk. On replay the RAM buffer is filled with bursts from the disc blocks and read out in a continuous form. In order to preserve the original order of samples, the buffer must operate in the First-In-First-Out (FIFO) mode.
- Timebase Correction - The timing of data entering the buffer may be erratic or have gaps. The timing of data leaving the buffer is made steady by using a reference clock to control the reading process.
- Synchronisation - buffers are used to synchronise audio data with an external reference such as Time Code by controlling the rate at which data is read out to ensure lock.
- Smooth Editing - Discontinuities of transitions between various sound files are smoothed out and made continuous at their join or cross-fade.

The Allocation Unit (AU)

A minimum AU is defined representing a package of contiguous blocks.

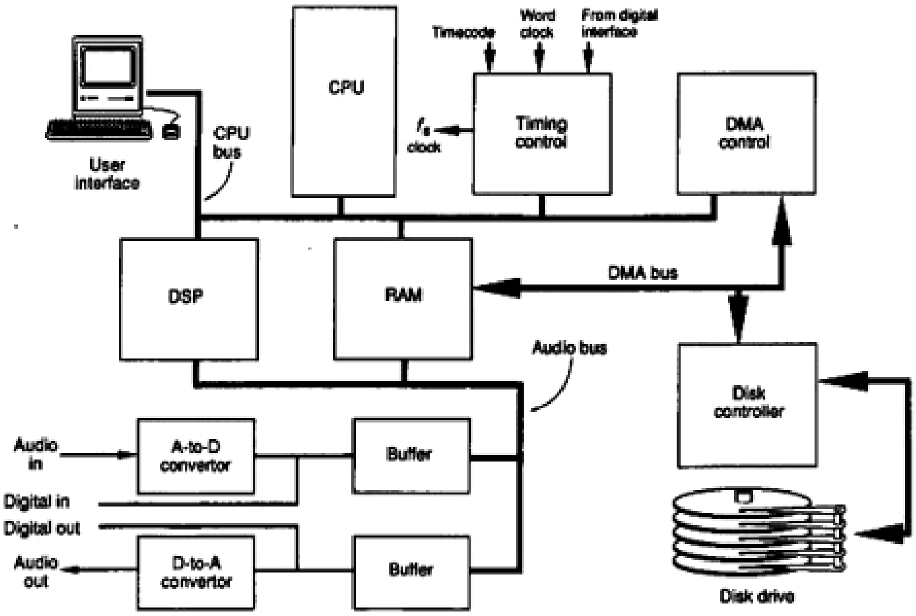
E.g. An AU of 8 Kbs = 16 x 512 bytes

Access Time (ms)

The time taken between the system requesting a file from the disk and the first byte of that file being accessed by the disc controller. Also access time can mean the time taken for the head to jump from file to file.

In a disk drive system access time is governed by the speed at which the read/write heads can move accurately from one place to another, and the physical size of the disc. The head must move radially across the disk

Tapeless sound recording



Conceptual block diagram of a generalized tapeless recording system. Audio is transferred to and from the store by DMA (direct memory access), perhaps via digital signal processing (DSP). Memory is used as a temporary store, and FIFO buffering is employed to smooth inputs and outputs. A user interface displays information and handles commands, communicating with the CPU. A timing interface handles synchronization

leading to a delay called seek latency. Then the disc rotates until the desired block moves under the head (rotational latency). Hence, average access time for a particular hard disk will be the sum of its seek and rotational latencies.

Transfer Rate (MB/sec)

The rate at which data can be transferred to and from the disk once the relevant location has been found. This is a measure of buss speed and CPU rate. Transfer rate in conjunction with access time limits the number of channels that can be successfully recorded or played back. These 2 factors also limit the freedom with which long crossfades and other operational features may be implemented.

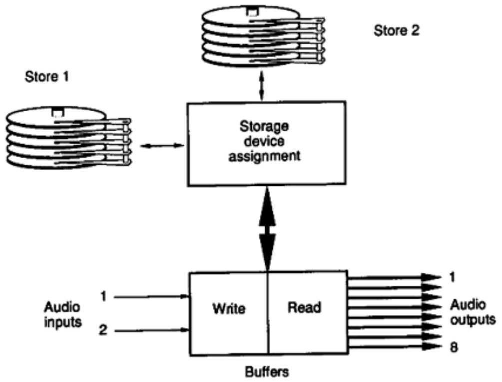
Disk Optimization

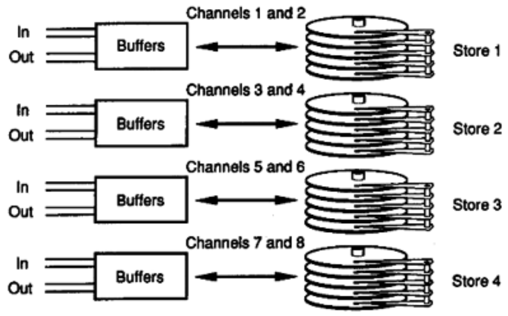
To get the best response out a hard disk recording system the efficiency of data transfer to and from the store must be optimised by keeping the number of access to a minimum for any given file.

7. Multichannel Considerations

In a tapeless system, the concept of track is very loosely defined and a “channel" refers to how many physical monophonic audio inputs and outputs there are in the system.

(a)





Multichannel disk recording systems often use more than one disk drive since there is a limit to the number of channels which can be serviced by a single drive. This leads to a number of considerations:

Determine how many channels a given disk can handle.

Next work out how many disks are required for the total storage capacity.

Decide which flies or channels should he written to which drives. A storage strategy which places files physically close to another based on their time contiguous relationship will favour a faster playback if these files are to be read off the disk in this same order. Hence a system which imitates a multitrack tape machine, perhaps assigning several channels per disk, is a good strategy, particularly where the playback of instruments in a music piece is required. In sound design where sound FX are randomly access according to picture considerations the is strategy may not work so well.

7.1 Winchester Magnetic Disc Drives

Used in PCs. Contained within a sealed unit to stop disk contamination.

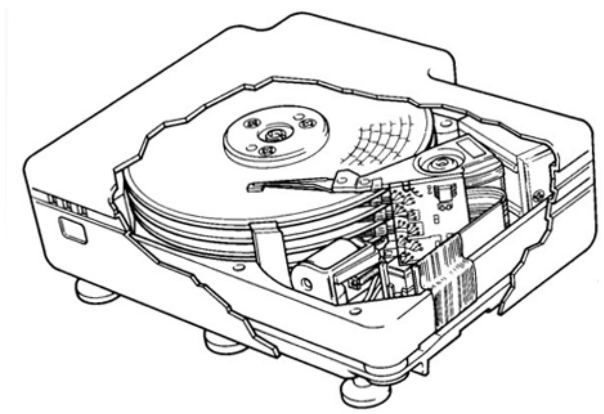


Figure 5.1 Inside a Winchester hard disk drive, a number of platters revolve at high speed on a common spindle. Heads are controlled by arms and a rotary positioner, one head for each surface. Data are recorded in concentric rings (tracks), subdivided into blocks. (Courtesy of MacUser magazine)

The disks are rigid platters that rotate on a common spindle. The heads 'float' across the surface, lifted by the aerodynamic effect of the air produce between the positioner and the disk rotation.
Data is stored in a series of concentric rings (tracks). Each track is divided up into blocks. Each block is separated by a small gap and preceded by an address mark which uniquely identifies the block location. The term Cylinder relates to all the tracks which reside physically in line with each other in the vertical plane, through the different disk surfaces. A Sector refers to a block projected onto the multiple layers of the cylinder.

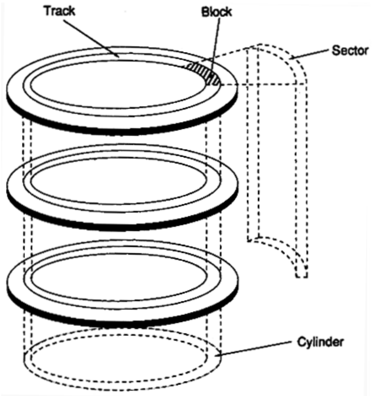


Diagram illustrates the difference between sector, block, track and cylinder in a hard disk's platters.

8. Editing in Tapeless Systems

Pre-recorded soundfiles are replayed in a predetermined sequence, in accordance with a replay schedule called an Edit Decision List (EDL). Memory buffering is used to smooth the transition from one file to another. Using non-destructive editing, any number of edited masters can be compiled from one set of source files, simply by altering the replay schedule. In this way edit points may be changed and new takes inserted without ever effecting the integrity of the original material.

Non-destructive Crossfade

When performing a short fade from file X to file Y. Both files are read out via memory. At the time of the cross fade the system ensures that data from both files exists simultaneously in different address areas, by reading from the disk ahead of the realtime playback. The exact overlap between old and new material will depend on a user-specified crossfade. At the start of the crossfade the system reads out both X and Y samples into a crossfade processor. Time coincident X and Y samples are blended together and sent to the appropriate channel output.

Because the system must maintain audio data simultaneously from two audio regions, the demands on memory are high. Thus, in general, the larger RAM capacity, the longer and more gradual the crossfade.

Destructive Crossfade

Can be made as long as the user wants but involve non-realtime calculation and separate storage of the crossfaded file. There are two variations of the destructive crossfade:

A real edited master recording is created from the assembled takes which would exist as a separate soundfile. You either require more disc space for this operation or you wipe over your previous flies.

Crossfade segments are created and stored separately from the main soundfiles. This saves on disc space but allows for long crossfades. This is not a realtime process and the user has to wait for the results.

Edit-point Searching

Often there is a user interface utilising a moving tape metaphor, allowing the user to cut, splice, copy and paste files into appropriate locations along a virtual tape. Variable speed replay in both directions is used to simulate 'scrubbing' or reel rocking.

The Edit Decision List (EDL)

The heart of real-time editing process is the EDL which is a list of soundfiles sent to particular audio outputs at particular times. The EDL controls the replay process and is the result of the operator, having chosen the soundfiles and the places - often specified in SMPTE addresses - at which they are to be joined. To achieve the final EDL the operator will have auditioned each soundfile and determined the crossfade points.

9. Pro-tools (Use manual)