Información Digital
Representación y Codificación

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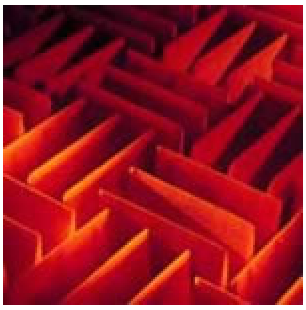
http://www.indiana.edu/~emusic/etext/digital_audio/chapter5_digital.shtml

Chapter Five: Digital Audio



8. Digital audio file formats

Digital audio files, which are the storable and editable collection of samples organized in a standard form, can be stored on computer drives, transferred to other computers or samplers, shared on the Internet to be downloaded or played-back in real time. They are different from audio CD or DAT tracks, which mostly contain only the raw sample data or items useless to a computer, such as error-correction and subcode data to help point the CD laser, etc. That is why a CD track must be "extracted" or "ripped" to an audio file format to be usable by a computer application. A standard 16-bit, 44.1K stereo file eats up about 10 megs of disk space per minute of sound.

Audio files come in a variety of types, which can influence their bit depth, multi-channel organization, compression scheme, sampling rate, organization of bytes high to low or visa versa (called "endian-ness") and amount of non-sample information stored in an area called the header, in units called chunks. Many audio programs are capable of opening and converting several file formats within limits. Some additional terms you may see when looking at a soundfile format is related to the bit-depth often tied to how computers store different sized numbers. Common sample sizes are often called 8-bit chars, 16-bit short integers, 32-bit unsigned long or 32-bit floating point (floats).

Stereo soundfiles can be organized as **interleaved**, where the sample bytes of respective channels alternate in a single stream (LRLRLR, etc), or as two separate files called **split stereo**, where one file contains the LEFT channel samples and another file contains the RIGHT channel samples. By convention, these are usually labeled with the same name with a .R or .L suffix (ex. myaudio.L, myaudio.R). Most programs will simultaneously open both files by default. Many programs, such as MOTU's Digital Performer and Digidesign's Pro Tools, work only with split stereo files—when importing an interleaved file, they will automatically split it into two files. However, some CD burning programs will burn only interleaved stereo files, so the separate files must be "bounced to disk" and then exported as an interleaved file to be burned.

Many of the file formats were designed to work with a specific processor chip. For example, the AIFF format was designed for the Motorola 680x0-based Apple Macintosh, which uses the Most Significant Byte (MSB) first and the Least Significant Byte (LSB) last (this is called Big Endian—it takes two 8-bit bytes to make a 16-bit sample), as opposed to the Microsoft WAVE format, which was designed for the Intel 80x86 processors, in which the LSB comes first (Little Endian), which is the way the processor handles most information, confirming some's belief that Intel thinks backwards (decimal '21' would be coded as '12' in Little Endian style).

Some of the common sound file format types are (.xxx indicates common filename extensions used for these formats):

- **.aif** or **AIFF** (Audio Interchange File Format), the gold standard of 16-bit audio, travels well between almost all computers and software, includes header information like file name, sampling rate, MIDI note number for samplers, loop points, number of bytes in file. Also capable of 24-bit and 32-bit resolution. Has the capability for quad, but rarely used. Click here for specs.
- **.aifc** or **AIFC** or AIFF-C (compressed version of AIFF—does not have to be compressed, supports both Big Endian, popular with SGI computers). Was thought to be candidate to supercede AIFF, since it had all the AIFF properties with more capabilities, but that has not happened as of this writing. Some AIFF-happy programs will choke on AIFC, particularly if compressed. Click here for a .pdf draft spec from Apple Computer.
- **.sd2** or **SD II** (Sound Designer II—same as AIFF with added proprietary information such as markers and regions—still very popular on Macs, even though Sound Designer is defunct). WARNING: not portable to non-Mac computers.
- **.mp3** (MPEG I-audio layer 3 compression—In 1987, the Fraunhofer IIS-A started to work on perceptual audio coding in the framework of the EUREKA project . In a joint cooperation with the University of Erlangen, the Fraunhofer IIS-A finally devised a very powerful algorithm that is standardized as ISO-MPEG Audio Layer-3 . With the proper codecs, compression rates of up to 24 times can be achieved with near- (but not) CD-quality. The beauty of MP3 is it's size vs. perceived quality, also its ability to be downloaded and then loaded into the flash memory of MP3 players. It can also be streamed to MP3 client software, recognized most Web browser audio helper applications. Files are encoded at certain bit-rates for target download speeds; for example, very good quality can be attained with 160 kbps encoding. Would you want to master all your music on MP3--no, but at least you can listen to it while you're jogging. Click here for massive information overload.
- **.ra** or **.ram** (Real Audio—can be streamed on the Internet from a Real Audio server, so sound starts playing before file fully downloaded. They can be encoded at multiple sampling rates to accommodate different user download speeds (modem, DSL, T1 lines, etc.) which range from 8 kbps to 1.5 Mbps (don't try 1.5 Mbps on your grandmother's 28.8 modem). Can also be combined with video for Real Media streaming. It spreads compression artifacts across the spectrum so they are theoretically not as noticeable. Requires Real Audio or compatible player software client. See www.real.com.
- **.wav** or Microsoft **WAVE** (Designed for PCs and Windows, but now usable with most audio programs, Mac or PC. Similar to AIFF for bit-depth and sample rates. As mentioned above, it uses MSB's and LSB's in reverse order of AIFF files, so Microsoft developed the RIFF interchange File Format to support the "Little Endian" scheme. Click here for specs).
- **WMA** or **Windows Media Audio** designed for use with Window Media Player with various compression ratios.
- **.au** (a-law, used with Sun computers or .snd used with NeXT's)
- **.ul** (u-law US telephony, headerless, usually 8-bit, low quality)
- **.sf** (IRCAM)
- all sorts of surround sound formats for encoding and decoding multiple channels of audio, often for video, film or DVD's (Dolby Pro-Logic, Dolby Digital Surround 5.1, THX, DTS, etc.. Beyond the scope of this article, but click here for further information. With the advent of home DVD burners, watch this take off as a viable audio medium.).
- Also watch for AAC, or Advanced Audio Coding Scheme being developed by Sony, ATT, Dolby Labs, and the original MP3 folks, which may encode multi-channel 5.1 surround files, as well as other mono, stereo other massive multi-channel formats at lower bit-rates up to 96 kbps and 24-bit resolution. It bypasses the limitations of mp3 in that is it

NBC (non-backward compatible), though it is based on the MPEG-2 standard. The new AAC MPEG-4 adds even further quality for coding at low bit rates and in fact it seems like AAC *is* the MPEG-4 audio standard. Click here for further information.

A fantastic and free (thanks Tom) program, SoundHack by Tom Erbe for converting soundfile formats and much more can be found at: www.soundhack.com. Also, thanks to the authors of other sites linked to in this article.