Music and Media
13 September 2004

<u>Lecture 2</u>

Hello class! Our digital music lab is in the process of being moved to the basement of the main building, next to the recording studio. While all the equipment has been safely stowed away, the workstations are not set up, and most of the machines need to be re-installed. Then there's the issue of being able to fit all of us in our new tiny room! I promised at the beginning of this semester that I would not unleash the class into an unfinished and malfunctioning lab environment, so we may not be working with the lab for a few weeks. The good news is, we've found all the missing headphones so you all can work in relative silence. ☺

The topic of our lecture today is on different formats of digital audio. Digital audio comes in many different formats, and multiple formats will be a fact of life for the foreseeable future. Groups like MPEG have created open standards, but even formats based on the same MPEG standard may not compatible with each other because of proprietary components.

Fortunately for consumers, many hardware and software players are able to support multiple formats—so if you purchase digital music in any of the major formats (MP3, WMA, etc.) you will be in a good shape. If a format does become obsolete, plenty of tools are available for converting digital audio to different formats.

Digital Audio Files

An audio file has two main parts: a header and the audio data. The header is used to store information about the file, including the resolution, sampling rate and type of compression. Often a "wrapper" is used to add features, such as license management information or streaming capability, to a digital audio file.

Figure 25 - Digital Audio File Structure

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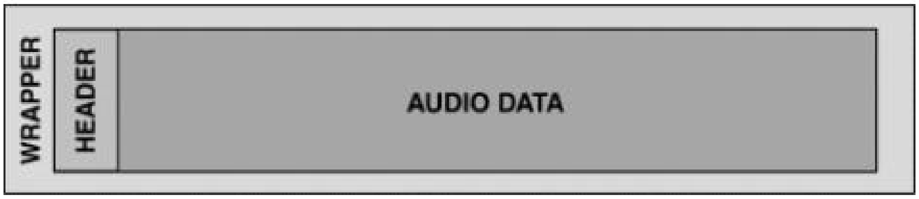


The format of a digital audio file refers to the type of audio data within the file. The file type refers to the structure of the data within the file. It is common for the same format to be used by more than one file type. For example, the PCM format is found in both WAV and AIFF files.

Table 1 - Common Digital Audio Formats

Type	Extensions	Codec
AIFF (Mac)	.aif, .aiff	*PCM
AU (Sun/Next)	.au	*u-law
CD audio (CDDA)	N/A	PCM
MP3	.mp3	MPEG Audio Layer-III
Windows Media Audio	.wma	Proprietary (Microsoft)
QuickTime	.qt	Proprietary (Apple Computer)
RealAudio	.ra, ram	Proprietary (Real Networks)
WAV	.wav	*PCM

* Can be used with other codecs.

WAV

WAV is the default format for digital audio on Windows PCs. WAV files are usually coded in PCM format, which means they are uncompressed and take up a lot of space. WAV files can also be coded in other formats, including MP3.

AIFF and AU

AIFF is the default audio format for the Macintosh, and AU is the default format for SUN systems. Both of these formats are supported on most other platforms and by most audio applications. Each of these formats can be compressed, but compression sometimes creates compatibility problems with other platforms.

Streaming Audio

Streaming audio avoids many of the problems of large audio files. Instead of having to wait for the entire file to download, you can listen to the sound as the data arrives at your computer.

Streaming audio players store several seconds worth of data in a buffer before beginning playback. The buffer absorbs the bursts of data as they are delivered by the Internet and releases it at a constant rate for smooth playback.

Many digital audio formats can be streamed by wrapping them in a streaming format, such as Microsoft ‘s ASF (Active Streaming Format), which can be used to stream MS Audio, MP3 and other formats.

Table 2 - Streaming Audio Systems

Type	Primary Format	Developer
Windows Media Technologies	Windows Media Audio / Active Streaming Format (ASF)	Microsoft
Icecast (open source)	MP3	The Icecast Team
QuickTime	QuickTime	Apple Computer
RealSystem	RealAudio	RealNetworks
SHOUTcast	MP3	Nullsoft

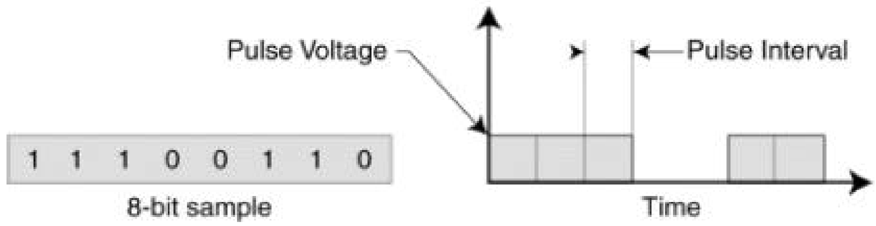
Standard Formats

Standard formats make it easier for software developers and equipment manufacturers to produce products that are less costly and more compatible with each other. The compatibility provided by standard formats helps assure consumers that their music and equipment won’t become obsolete. Cassette tapes, compact discs and PCM are examples of standard audio formats that benefit both consumers and manufacturers.

PCM

PCM (Pulse Code Modulation) is a common method of storing and transmitting uncompressed digital audio. Since it is a generic format, it can be read by most audio applications—similar to the way a plain text file can be read by any word-processing program. PCM is used by Audio CDs and digital audio tapes (DATs). PCM is also a very common format for AIFF and WAV files.

Figure 26 - Pulse Code Modulation



PCM is a straight representation of the binary digits (1s and 0s) of sample values. When PCM audio is transmitted, each “1” is represented by a positive voltage pulse and each “0” is represented by the absence of a pulse. Figure 26 shows how binary data is converted to a PCM signal.

DPCM

DPCM (Differential Pulse Code Modulation) is a simple form of lossy compression that stores only the difference between consecutive samples. DCPM uses 4 bits to store the difference, regardless of the resolution of the original file. With DCPM, an 8-bit file would be compressed 2=1, and a 16-bit file would be compressed 4=1.

ADPCM

ADPCM (Adaptive Differential Pulse Code Modulation) is similar to DCPM except that the number of bits used to store the difference between samples is varied depending on the complexity of the signal. ADPCM works by analyzing a succession of samples and predicting the value of the next sample. It then stores the difference between the calculated value and the actual value.

u-law Compression

u-law (pronounced “mew-law”) is a common lossy compression scheme, similar to ADPCM, which can be used on AU, AIFF and WAV files.

MPEG Audio

MPEG Audio is a family of open standards for compressed audio that includes MP2, MP3 and AAC. (See Chapter 13 for more detailed information on MPEG Audio.)

Apple QuickTime

QuickTime is a widely used multimedia format from Apple Computer that supports both streaming audio and streaming video. Much of the MPEG-4 standard is based on QuickTime, and it is widely used for streaming video on the Web.

Non-MPEG Proprietary Formats

Several digital audio formats exist that are entirely proprietary. Many of these are quite good and are widely used.

Dolby Digital (Formerly AC-3)

Dolby Digital is a very high quality audio encoding and noise reduction system that is the audio component of High Definition Television (HDTV) and digital broadcast TV (DTV). It is also used in DVDs, laser discs, digital cable and direct broadcast satellite (DBS) systems.

EPAC

EPAC is a perceptual audio encoding scheme based on PAC—developed by Bell Labs, the research and development arm of Lucent Technologies. EPAC is reported to produce quality indistinguishable from the original CD at 128 kbps. However, I participated in one listening test where the audience was able to consistently tell the difference between original CD tracks and the same tracks encoded in EPAC at 160 kbps.

Windows Media Audio

Microsoft's Windows Media Audio (WMA) format is a relatively late entry into the field of proprietary audio formats. WMA performs very well at lower bit-rates and is reported to produce quality indistinguishable from the original CD at 128 kbps. WMA is supported by most full-featured player programs and by many portable players. WMA is royalty-free when incorporated into software that runs on the Windows platform.

RealAudio

RealAudio was the first widely used system for streaming audio and video over the Internet. It is a proprietary format, but it is used by many online music stores for sample clips of songs. The RealPlayer also provides support for MP3.

TAC

TAC (Transparent Audio Compression) is a high-quality perceptual encoding scheme developed by K+K Research. TAC uses Adaptive Bit-rate Management (ABM), which is similar to VBR (variable bit-rate) encoding. TAC was developed as part of K+K Research's MP02 (Music Publisher 02) software.

TwinVQ (VQF)

TwinVQ (Transform-domain Weighted Interleave Vector Quantization) is an encoding scheme developed by the NTT Human Interface Lab in Japan. TwinVQ is reported to provide higher quality than MP3, but encoding times are reported to be much longer, and CPU utilization is reported to be higher during playback.

Homework 1

Our first assignment will be taking place over three weeks. It is an introduction to basic audio editing software, essentially a simplified version of what you will be using in the lab once it is up and running.

We will be using a freeware (as in, free of charge to distribute and use) audio editor called **Audacity**. You can download it by going to the class website at **http://www.solscope.com/~ochsa/** and selecting the link there, or by visiting the official site at **http://audacity.sourceforge.net/**. It runs on computers with Windows, Macintosh, or Linux operating systems. If you do not have access to the Internet, please let me know and I will provide the software for you on physical media. If you do not have a computer at home, please let me know so that I can provide one for you to use for the duration of this assignment.

This first week, your assignment is to:

- Download the software
- Download the sample audio files from the class webpage
- Install the software on your computer
- Load the sample audio files into Audacity and play them back

If you can get to this stage, you will be prepared for part two of this homework, in which we will learn various processing techniques for editing digital audio (next week). If you get stuck somewhere in this process, please email **ochsa@solscope.com**.

You might be wondering why any of this is important. If you ever plan on recording your own playing, and/or applying for a music school/summer festival/etc., then being able to create your own recordings is an essential skill. The tools to make an audio CD are freely available this day in age, and the overall cost is low enough such that any student needing to produce a recording should be able to get a hold of the resources to do so. The process is fairly simple, and this knowledge is valuable to you as a musician in the future!