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WAV File Format

Introduction

A WAV file is typically used to store raw uncompressed audio data. The WAV file format is a subset of the RIFF file format which can be used to store any type of data using what are known as "chunks". WAV files can also be used to store any "streaming" type data (not just audio). WAV files are often used to record music since the uncompressed audio data is "lossless"- as opposed to compressed audio formats such as MP3 which are "lossy". The downside of the WAV audio format is the files can be huge. For example, when recording "CD quality" audio, which is stereo (i.e. two channel) 16 bit samples at a sample rate of 44.1Khz, there will be 176,400 bytes of data stored for every second of audio. Therefore, a four minute long piece of music would require a file size of over 42MBytes.

WAV files can also be used to store compressed audio and can include "meta data" such as the song title, artist name, etc. WAV files might also include extra data "chunks" which might be used to indicate specific locations in the audio data stream, etc. This document only details the most common WAV file format for uncompressed audio with no meta data which samples the audio in linear pulse code modulation format (LPCM). LPCM format records raw audio data directly from the A-to-D converter at a fixed sample rate. The characteristics of this type of WAV file are:

1. The WAV file header is 44 bytes.
2. String data is straight ASCII.
3. All other data is in little endian format.
4. Raw sample data directly follows the 44 byte header.
5. Sample data = 8 bit is unsigned.
   Sample data = 16 bit is signed.
6. There are only three data "chunks" in the file:
   the RIFF header, the audio format chunk and the actual audio samples data chunk.

The example below shows the common WAV file format.

Table 1: WAV File Format Example

	Num Bytes	Example Data	Description
44 Byte Header	4	0x52,0x49,0x46,0x46	"RIFF"
	4	0xC4,0x0F,0x00,0x00	file size - 8 = 4036
	4	0x57,0x41,0x56,0x45	"WAVE"
	4	0x66,0x6D,0x74,0x20	"fmt "
	4	0x10,0x00,0x00,0x00	format chunk size = 16
	2	0x01,0x00	audio format (1 = LPCM uncompressed)
	2	0x02,0x00	num channels = 2
	4	0x40,0x1F,0x00,0x00	sample rate = 8000
	4	0x00,0x7D,0x00,0x00	data rate bytes/sec = 32000
	2	0x04,0x00	num channels * bytes/sample = 4
	2	0x10,0x00	bits/sample = 16
	4	0x64,0x61,0x74,0x61	"data"
	4	0xA0,0x0F,0x00,0x00	num samples * num channels * bytes/sample
Raw Data	2	0xA7,0xCC	channel 1, sample 1
	2	0x50,0x27	channel 2, sample 1
	2	0x5A,0x2D	channel 1, sample 2
	2	0xE9,0x2E	channel 2, sample 2
	2	0xCC,0xC5	channel 1, sample 3
	2	0xA9,0xBA	channel 2, sample 3
	2	0x29,0xEE	channel 1, sample 4
	2	0xCE,0x22	channel 2, sample 4
		- - - etc - - -	

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