



مرکز آموزش نیرا سیستم

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Audio

Ali Mirghasemi



What is an Audio File?

- An audio file is a digital recording or representation of sound.
- It contains information about the amplitude and frequency of sound waves over time.
- Audio files are used for various purposes, including music, speech, sound effects, and more.
- Different audio file formats exist to store and transmit audio data efficiently.



Audio Codec

- **Lossy Codecs**

- These codecs discard some audio data during compression, resulting in a smaller file size
- **Ex:** MP3, AAC, WMA

- **Lossless Codecs**

- Lossless codecs compress audio data without losing any information, allowing for perfect reconstruction of the original audio
- **Ex:** FLAC, ALAC, WAV (in some cases)



Application

- **Streaming Services**

- Codecs like AAC and MP3 are commonly used for streaming audio over the internet due to their efficient compression.

- **Music Storage**

- Lossless codecs like FLAC and ALAC are favored for storing high-quality music files on devices with abundant storage.

- **Voice Communication**

- Codecs like Opus are optimized for real-time communication applications, such as VoIP and online gaming.

- **Broadcasting**

- Codecs like MPEG-2 Layer III (MP3) are used for radio broadcasting, allowing for the transmission of audio with reduced bandwidth requirements.



WAV File

- Waveform Audio File Format (WAV) is a popular uncompressed audio file format developed by Microsoft and IBM.
- WAV files contain raw audio data without any compression, resulting in high audio quality but larger file sizes.
- WAV files are commonly used in professional audio production, where preserving the original quality is crucial.

WAV Header

endian	File offset (bytes)	field name	Field Size (bytes)	
big	0	ChunkID	4	The "RIFF" chunk descriptor
little	4	ChunkSize	4	
big	8	Format	4	
big	12	Subchunk1 ID	4	The "fmt" sub-chunk
little	16	Subchunk1 Size	4	
little	20	AudioFormat	2	
little	22	NumChannels	2	
little	24	SampleRate	4	
little	28	ByteRate	4	
little	32	BlockAlign	2	
little	34	BitsPerSample	2	
big	36	Subchunk2ID	4	The "data" sub-chunk
little	40	Subchunk2 Size	4	
little	44	data	Subchunk2Size	

The "RIFF" chunk descriptor

The Format of concern here is "WAVE", which requires two sub-chunks: "fmt " and "data"

The "fmt " sub-chunk describes the format of the sound information in the data sub-chunk

The "data" sub-chunk Indicates the size of the sound information and contains the raw sound data



PCM (Pulse Code Modulation)

- PCM is a method used to digitally represent analog signals, including audio.
- It involves sampling the amplitude of an analog signal at regular intervals and converting each sample into a digital code.
- PCM is the basis for many audio file formats, including WAV.
- PCM can be either linear or nonlinear, with linear PCM being the most common.



How to Play Audio

- **Audio File Decoding**

- The microcontroller needs to decode the audio file, especially if it's in a compressed format. This may require using a dedicated audio codec or a software decoder.

- **Digital-to-Analog Conversion (DAC)**

- The microcontroller must convert the digital audio data into analog signals that can be played through speakers or headphones. This is achieved using a DAC.

- **Amplification**

- In many cases, the analog signal may need amplification before being sent to speakers to ensure sufficient volume.

- **Output to Speakers**

- Connect the output from the DAC or amplifier to speakers or headphones for audio playback.