

NOTES

Bachelor of Science (Third Year)

CSD-106-Multimedia Techniques

Title of the Unit : Sound

Module Name : Audio Codec & File formats, processing sound

Module Number : 18

A codec performs the encoding and decoding of the raw audio data while this encoded data is (usually) stored in a container file. In hardware, audio codec refers to a single device that encodes analog audio as digital signals and decodes digital back into analog. In other words, it contains both an analog-to-digital converter (ADC) and digital-to-analog converter (DAC) running off the same clock signal. In software, audio codec is a computer program implementing an algorithm that compresses and decompresses digital audio data according to given audio file or streaming media audio coding format.

Although most audio file formats support only one type of audio coding data (created with an audio coder), a multimedia container format (as Matroska or AVI) may support multiple types of audio and video data. Common Audio codecs are AAC and MP3.

Audio File Formats:

An audio file format is a file format for storing digital audio data on a computer system. The bit layout of the audio data (excluding metadata) is called the **audio coding format** and can be uncompressed, or compressed to reduce the file size, often using lossy compression. The data can be a raw bitstream in an audio coding format, but it is usually embedded in a container format or an audio data format with defined storage layer. There are three major groups of audio file formats:

Uncompressed audio formats: WAV, AIFF, AU or raw header-less PCM

WAV and AIFF are designed to store a wide variety of audio formats, lossless and lossy; they just add a small, metadata-containing header before the audio data to declare the format of the audio data, such as LPCM with a particular sample rate, bit depth, endianness and number of channels.

WAV and AIFF are widely supported and can store LPCM, they are suitable file formats for **storing and archiving an original recording**.

PCM –It stands for Pulse-Code Modulation. It represents raw analog audio signals in digital form. To convert analog signal into digital signal it has to be recorded at a particular interval. Hence it has sampling rate and bit rate (bits used to represent each sample). It is an exact representation of the analog sound and does not involve compression. It is the most common audio format used in CDs and DVDs

WAV –It stands for Waveform Audio File Format. It is just a Windows container for audio formats. That means that a WAV file can contain compressed audio. Most WAV files contain uncompressed audio in PCM format. It is just a wrapper.

AIFF –It stands for Audio Interchange File Format developed by Apple. AIFF files can contain multiple kinds of audio. It contains uncompressed audio in PCM format. It is just a wrapper for the PCM encoding.

Formats with lossless compression: FLAC, Monkey's Audio (filename extension .ape), WavPack (filename extension .wv), TTA, ATRAC Advanced Lossless, ALAC (filename extension .m4a), MPEG-4 SLS, MPEG-4 ALS, MPEG-4 DST, Windows Media Audio Lossless (WMA Lossless), and Shorten (SHN).

Lossless Compressed Audio Formats include FLAC, WavPack, Monkey's Audio, ALAC (Apple Lossless) A lossless compressed audio format stores data in less space without losing any information. The original, uncompressed data can be recreated from the compressed version.

Uncompressed audio formats encode both sound and silence with the same number of bits per unit of time. Encoding an uncompressed minute of absolute silence produces a file of the same size as encoding an uncompressed minute of music. In a lossless compressed format, however, the music would occupy a smaller file than an uncompressed format and the silence would take up almost no space at all. They provide a compression ratio of about 2:1 (i.e. their files take up half the space of PCM). Development in lossless compression formats aims to **reduce processing time while maintaining a good compression ratio.**

FLAC – It stands for Free Lossless Audio Codec. It can compress a source file by up to 50% without losing data. It is most popular in its category and is open-source.

ALAC – It stands for Apple Lossless Audio Codec. It was launched in 2004 and became free after 2011. It was developed by Apple.

WMA – It stands for Windows Media Audio. But it is least efficient in term of compression and is not open-source. It has limited hardware support.

Formats with lossy compression: Opus, MP3, Vorbis, Musepack, AAC, ATRAC and Windows Media Audio Lossy (WMA lossy). Opus, MP3, Vorbis, Musepack, AAC, ATRAC and Windows Media Audio Lossy (WMA lossy)

Lossy audio format enables even greater reductions in file size by removing some of the audio information and simplifying the data. This, of course, results in a reduction in audio quality

A variety of techniques are used, to remove the parts of the sound that have the least effect on perceived quality, and to minimize the amount of audible noise added during the process.

Most formats offer a range of degrees of compression, generally measured in bit rate. The lower the rate, the smaller the file and the more significant the quality loss.

Used to share audio files over network where **smaller file size matters over quality.**

MP3 – It stands for MPEG-1 Audio Layer 3. It was released in 1993 and became popular. It is most popular audio format for music files. Main aim of MP3 is to remove all those sounds which not hearable or less noticeable by humans ears. Hence making size of music file small. MP3 is like universal format which is compatible almost every device.

AAC – It stands for Advanced Audio Coding. It was developed in 1997 after MP3. The compression algorithm used by AAC is much more complex and advanced than MP3, so when compared a particular audio file in MP3 and AAC formats at the same bitrate, the AAC one will generally have better sound quality. It is the standard audio compression method used by YouTube, Android, iOS, iTunes, and PlayStations.

WMA – It stands for Windows Media Audio. It was released in 1999. It was designed to remove some of the flaws of MP3 compression method. In terms of quality it is better than MP3. But is not widely used.

MIDI

A piece of music could be shared in two ways: We could play it, record the performance, and send the recording, or we could write it down using some form of notation, indicating the arrangement, and send the sheet music, so you could play the piece for yourself. There is a parallel here with bitmapped and vector graphics. In the first case, we send you the actual sound, in the second, we send you what amounts to a set of instructions telling you how to produce the sound.

MIDI (The Musical Instruments Digital Interface) provides a way of delivering instructions about how to produce the music, that can be interpreted by suitable software or hardware. MIDI was devised as a standard protocol for communicating between electronic instruments, synthesizers, samplers, and drum machines.

Playing back MIDI files requires an instrument that understands MIDI, but a computer, equipped with suitable hardware or software, can be such an instrument itself. MIDI files are therefore a means of communicating music. Because they do not contain any audio data, they can be much more compact than actual digitized sound files.

MIDI messages

MIDI message is an instruction that controls performance of an instrument. A status byte indicates the type of the message, followed by one or two data bytes giving the values of parameters 'Note On', which takes two parameters: a number between 0 and 127 indicating the note to be sounded, and a key velocity, indicating how fast the key was pressed, and hence the attack of the note. 'Note Off', which ends a note, 'Key Pressure', which indicates the degree of 'after touch' to be applied, and 'Pitch Bend', to change note values dynamically

Processing sound

With the addition of suitable audio input, output and processing hardware and software, a desktop computer can perform the functions of a modern multi-track recording studio. Such professional facilities are expensive and demanding on resources. There is presently no single sound application that has a cross-platform desktop standard, like Photoshop and Premiere. Most of the well known ones are biased towards music, with integrated support for MIDI sequencing and multi-track recording.

Recording sound Where sound quality is important, or for recording music to a high standard, it will be necessary to use a properly equipped studio. Although a computer can form the basis of a studio, it must be augmented with microphones and other equipment in a suitable acoustic environment. Built-in microphones are not adequate for recording sounds as they will not give satisfactory results. Plugging an external microphone into a sound card, or doing the actual recording onto DAT (or good quality analogue tape) using a professional microphone, and capture it in a separate operation.

Before recording, it is necessary to select a sampling rate and sample size. Where the sound originates in analogue form, the choice will be determined by considerations of file size and bandwidth, which will depend on the final use. As a general rule, the highest possible sampling rate and sample size should be used, to minimize deterioration of the signal when it is processed.

The most vexatious aspect of recording is getting the levels right. If the level of the incoming signal is too low, the resulting recording will be quiet, and more susceptible to noise; if the level is too high, clipping will occur, that is, at some points, the amplitude of the incoming signal will exceed the maximum value that can be recorded.

Importing sound from an audio CD is a simpler alternative to recording sound.

Copyright-free collections can be obtained, much like royalty-free image libraries, although they tend to be anodyne.

Composers and musicians with access to professional recording facilities may supply their work on CD, avoiding the need for the multimedia producer to deal with the sound recording process. However, even when importing sounds from CDs there can be difficulty in getting the levels right.

Downloading files from Internet and listening to them may be legal, but using them in any published form without obtaining clearance from the copyright holders is illegal.