

## Digital Audio

Digital audio is the method of representing sound in digital form.

### Sound

- Sounds are pressure waves of air or other materials – there is no sound in the vacuum.



Figure 1: Sound Waves (source: <http://4.bp.blogspot.com/>)

- The speed of changes = *frequency* – measured in Hz
- People can hear sounds with the frequency between 16 Hz – 20 000 Hz

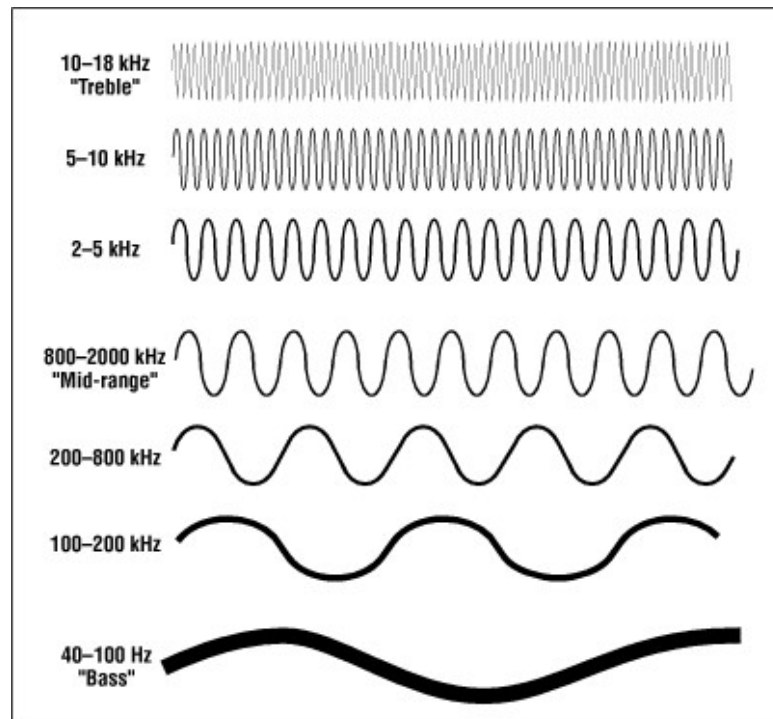


Figure 2: Audio Frequencies (source: <http://docstore.mik.ua/orelly/web2/audio/figs/aud.0207.gif>)

**Sound level (volume)** – energy of the sound pressure ○ measured in decibels (dB)

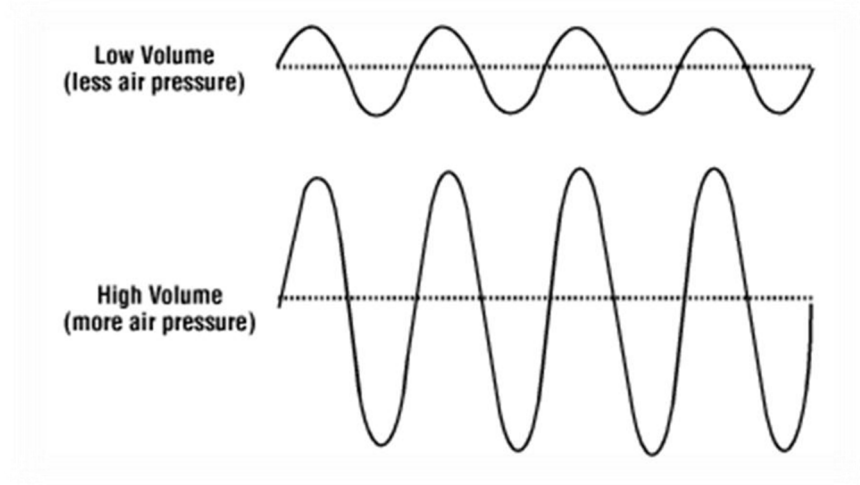


Figure 3: Sound Volume – two sound waves of the same frequency, but different volume (source: <http://docstore.mik.ua/oreilly/web2/audio/figs/aud.0204.gif>)

**Musical Tone** – periodic sound of a given frequency, duration, loudness, and timbre

**Timbre** – a quality of the tone, which is unique for different sources – e.g. timbre of a guitar, timbre of a violin

## Digital Audio Encoding

- **Sound – analog** = continuous wave with (theoretically) infinite number of values in both domain (time) and range (amplitude)
- **Computers** cannot store infinite amounts of data – they can **store approximate version of the sound** only.

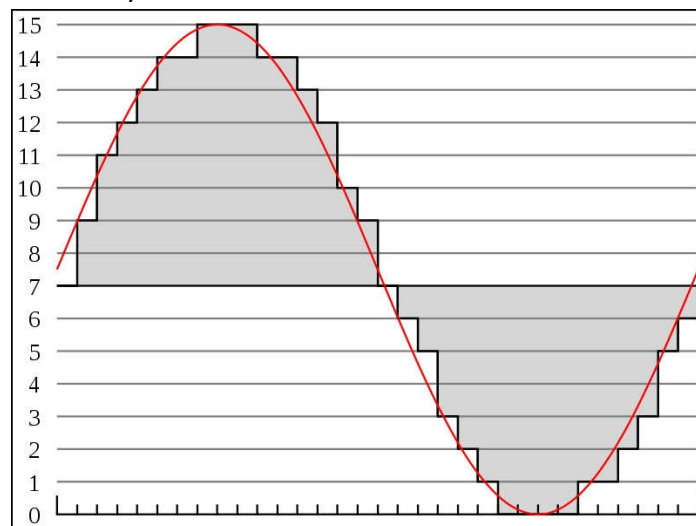


Figure 4: Original analog sound (red) and its digital approximation (black) (source: wikipedia.org)

- The creation of the digital version – **digitization** – in two phases:
  1. **Sampling**
  2. **Quantization**

## Sampling – Phase 1

- Regular measuring of the sound level ○
  - Regular – at a given frequency - **sampling rate** (in Hz) ○ Sound may contain multiple channels (2 channels for stereo or more for surround sound) – each should be sampled individually.

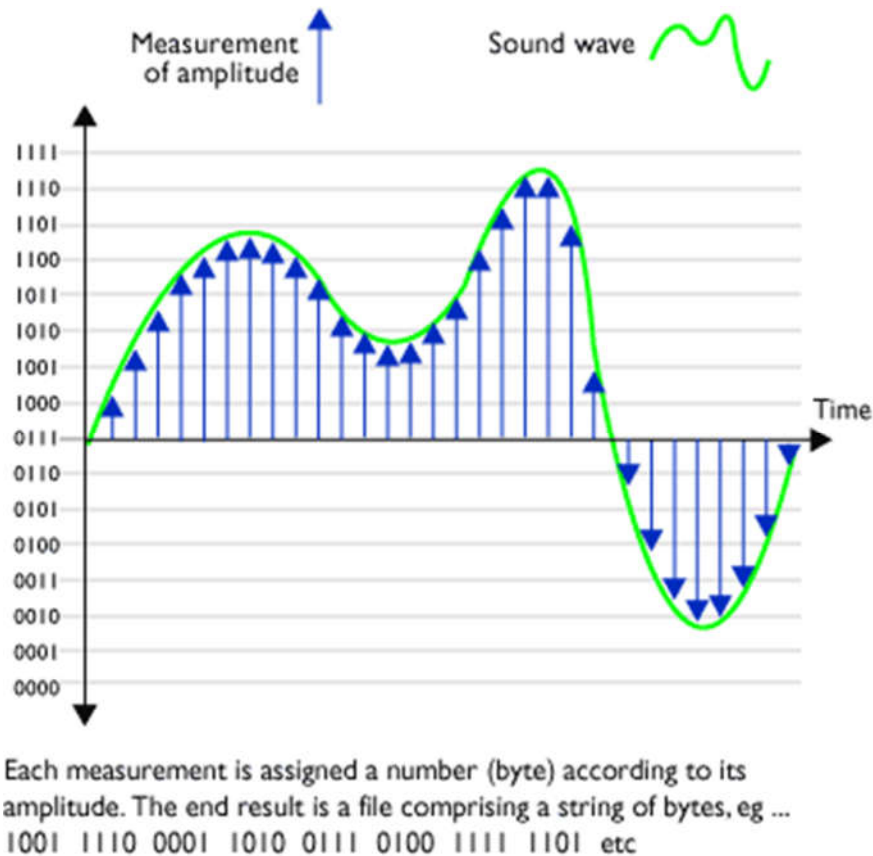


Figure 5: Sampling (source: [http://www.planetoftunes.com/digital-audio/how-do-analogue-to-digital-converters-work.html#.WM\\_chfnym8](http://www.planetoftunes.com/digital-audio/how-do-analogue-to-digital-converters-work.html#.WM_chfnym8))

- Generally speaking: **the higher the sampling rate and bit resolution the more fidelity, as well as increase the amount of digital data.**
- The original signal can be reproduced from the samples. **Nyquist–Shannon sampling theorem** states:  
*Faithful reproduction is only possible if the sampling rate is higher than twice the highest frequency of the signal .*
- *Example:* human sense of hearing has the top limit 20 000 Hz – to get acceptable quality of the digital sound the sampling rate should be 40 000 Hz at least (CD employs 44,1 kHz, professional audio has 48 kHz, 96 kHz, or even 192 kHz).

## Quantization – Phase 2

- Change of continuous amplitude levels to the discrete values
- The number of possible values is limited by the resolution
- **Resolution**
  - number of values, which can replace the measured amplitude
  - computers – store the results in bytes or group of bytes

Number of bits	Number of different values
1	2
2	4
3	8
8	256
16	65 536
24	16 777 216

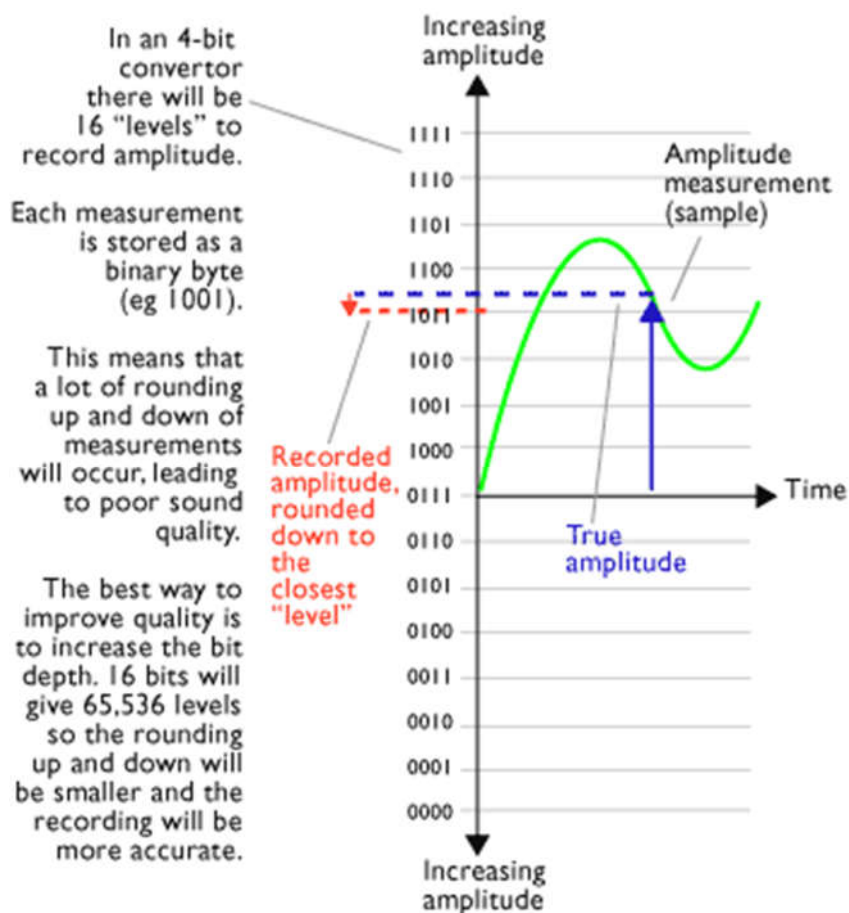


Figure 6: Quantization and Quantization Errors  
([http://www.planetoftunes.com/digital-audio/how-do-analogue-to-digital-converters-work.html#.WM\\_chfnyuM8](http://www.planetoftunes.com/digital-audio/how-do-analogue-to-digital-converters-work.html#.WM_chfnyuM8))

## Analog-To-Digital / Digital-to-Analog Converter (AD/DA C)

- a device, which converts audio to its digital form and vice versa
- a part of the sound card, digital recorder

## Audio file format

An audio file format is a container format for storing audio data on a computer system.

This data can be stored

- uncompressed or
- compressed to reduce the file size.

## Types of formats

There are three major groups of audio file formats:

1. **Uncompressed audio formats**, such as WAV, AIFF;
2. **Formats with lossless compression**, such as FLAC, Monkey's Audio, TTA, Apple Lossless and lossless Windows Media Audio (WMA);
3. **Formats with lossy compression**, such as MP3, Ogg Vorbis, lossy Windows Media Audio (WMA) and AAC.

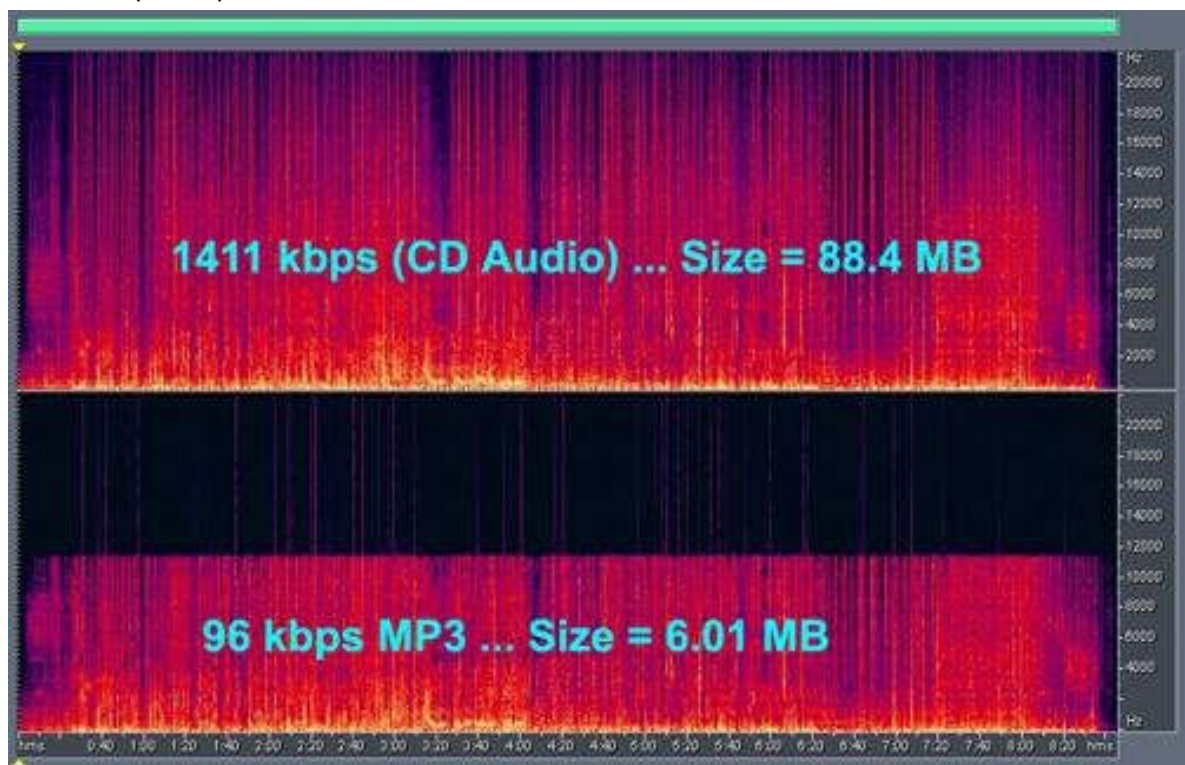


Figure 7: Comparison of the frequencies - uncompressed CD and lossy MP3 (source: <http://losslessmusic.org/faq>)

## Uncompressed audio format

There is one major uncompressed audio format, PCM, which is usually stored as a **.wav** on Windows or as **.aiff** on Mac OS. WAV is a flexible file format designed to store more or less any combination of sampling rates or bitrates. This makes it an adequate file format for storing and archiving an original recording. WAV, like any other uncompressed format, encodes all sounds, whether they are complex sounds or absolute silence, with the same number of bits per unit of time.

*As an example, a file containing a minute of playing by a symphonic orchestra would be the same size as a minute of absolute silence if they were both stored in WAV.*

## Lossless audio formats

Lossless audio formats (such as TTA and FLAC) provide a compression ratio of about 2:1 without discarding any samples. The cost of this losslessness is that the compression ratio is not good.

## Lossy Audio Formats

Lossy compression typically achieves far greater compression than lossless compression (data of 5 percent to 20 percent of the original stream, rather than 50 percent to 60 percent), by discarding less-critical data – those, which are not perceived by human auditory system.

## Open file formats

### Wav

standard audio file container format used mainly in Windows PCs. Commonly used for storing uncompressed (PCM), CD-quality sound files, which means that they can be large in size - around 10 MB per minute. Wave files can also contain data encoded with a variety of codecs to reduce the file size.

### Ogg

It is a free, open source format supporting a variety of codecs, the most popular of which is the audio codec Vorbis. Vorbis offers better compression than MP3 but is less popular.

### FLAC

It is a lossless compression codec. You can think of lossless compression as like zip but for audio. If you compress a PCM file to FLAC and then restore it again it will be a perfect copy of the original. Flac is recommended for archiving PCM files where quality is important.



## Proprietary formats

### mp3

The MPEG Layer-3 format is the most popular format for downloading and storing music. By eliminating portions of the audio file that are essentially inaudible, mp3 files are compressed to roughly one-tenth (1/10) the size of an equivalent PCM file while maintaining good audio quality.

### wma

It is a popular Windows Media Audio format owned by Microsoft.

### mp4

A proprietary version of AAC in MP4 with Digital Rights Management developed by Apple for use in music downloaded from their iTunes Music Store.

## Programs

### WinAmp

- It is a proprietary media player written by Nullsoft.
- It is skinnable, multi-format freeware / shareware.
- WinAmp was first released by Justin Frankel in 1997.
- In 2005 WinAmp grew from 33 million monthly users to over 57 million monthly users.
- Besides MP3, WinAmp supports a very wide variety of contemporary and specialized music file formats, including MIDI, MOD, MPEG-1 audio layers 1 and 2, AAC, M4A, FLAC, WAV and Windows Media Audio. WinAmp was one of the first common music players on Windows to support playback of Ogg Vorbis by default. It supports *gapless playback* for MP3 and AAC. In addition, WinAmp can play and import music from audio CDs, optionally with CD-Text, and can also burn music to CDs.

### Audacity

- Audacity is a free digital audio editor application. Audacity is cross-platform.
- Some of Audacity's features include:
  - Importing and exporting WAV, AIFF, MP3 (via the LAME encoder), Ogg Vorbis, and Free Lossless Audio Codec (FLAC)
  - Recording and playing sounds
  - Editing via Cut, Copy, Paste (with unlimited Undo)
  - Multi-track mixing
  - A large array of digital effects and plug-ins.
  - Amplitude envelope editing
  - Noise removal

- Support for multi-channel modes with sampling rates up to 100 kHz with 24 bits per sample
- The ability to make precise adjustments to the audio's speed, while maintaining pitch, in order to synchronize it with video, run for the right length of time, etc. (in primitive programs like Sound Recorder in Windows Accessories you can also change the speed, but it will shift also frequencies of the sound – in case of higher speed all frequencies are increased and vice versa)
- Multi-platform = it works on Windows, Mac OS X, and Unix-like systems (including GNU/Linux and BSD).



## Wordstock

<b>encoding</b>	kódovanie
<b>sampling rate</b>	vzorkovacia frekvencia
<b>resolution</b>	rozlíšenie
<b>analog-to-digital converter</b>	analógovo-digitálny konvertor
<b>continuous</b>	kontinuálny, plynulý, neprerušený
<b>discrete</b>	diskrétny, nespojitý, pozostávajúci s jednotlivých častí
<b>uncompressed</b>	nekomprimovaný
<b>compressed</b>	skomprimovaný
<b>lossless</b>	bezstratový
<b>lossy</b>	stratový
<b>proprietary</b>	proprietárny, patentovaný
<b>skinnable</b>	skinovateľný (majúci schopnosť meniť svoj vzhľad)
<b>gapless playback</b>	plynulé prehrávanie (bez pauzy medzi nahrávkami)
<b>amplitude envelope</b>	obal amplitúdy (umožňuje plynulo meniť amplitúdu v danom bode a jeho okolí)