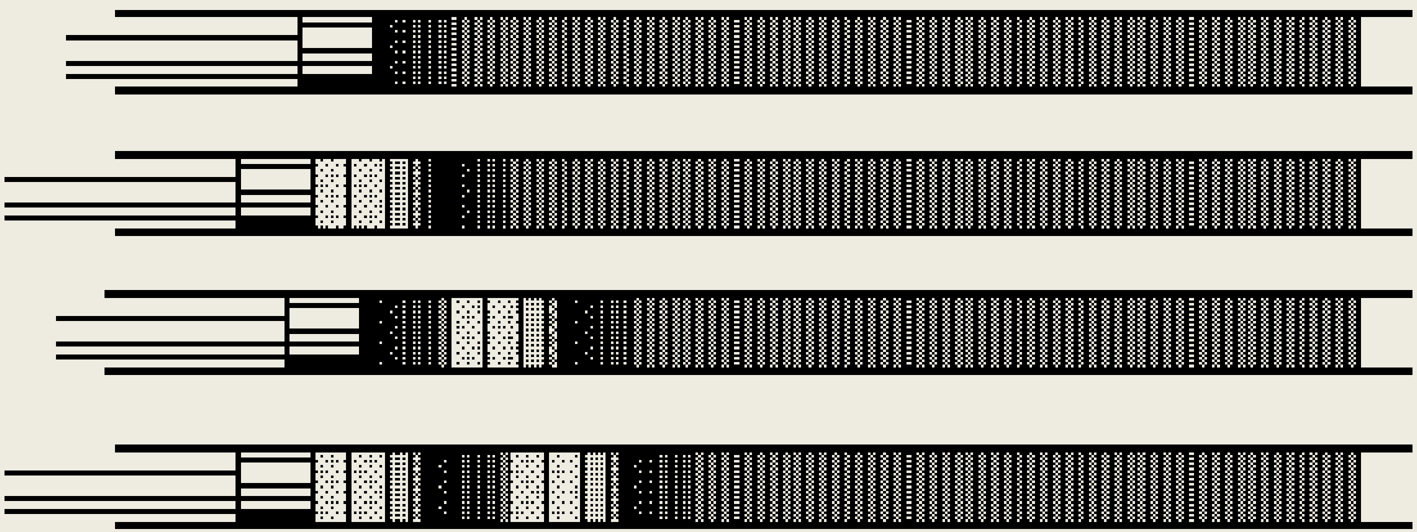
Audio

Acoustics : study of sound

That is: generation, transmission and reception of sound waves.

Basic Sound Concepts

- Sound is produced by vibration of material
    - During vibration, pressure variations are created in the surrounding air molecules.
    - Pattern of oscillation creates a waveform
        - the wave is made up of pressure differences.



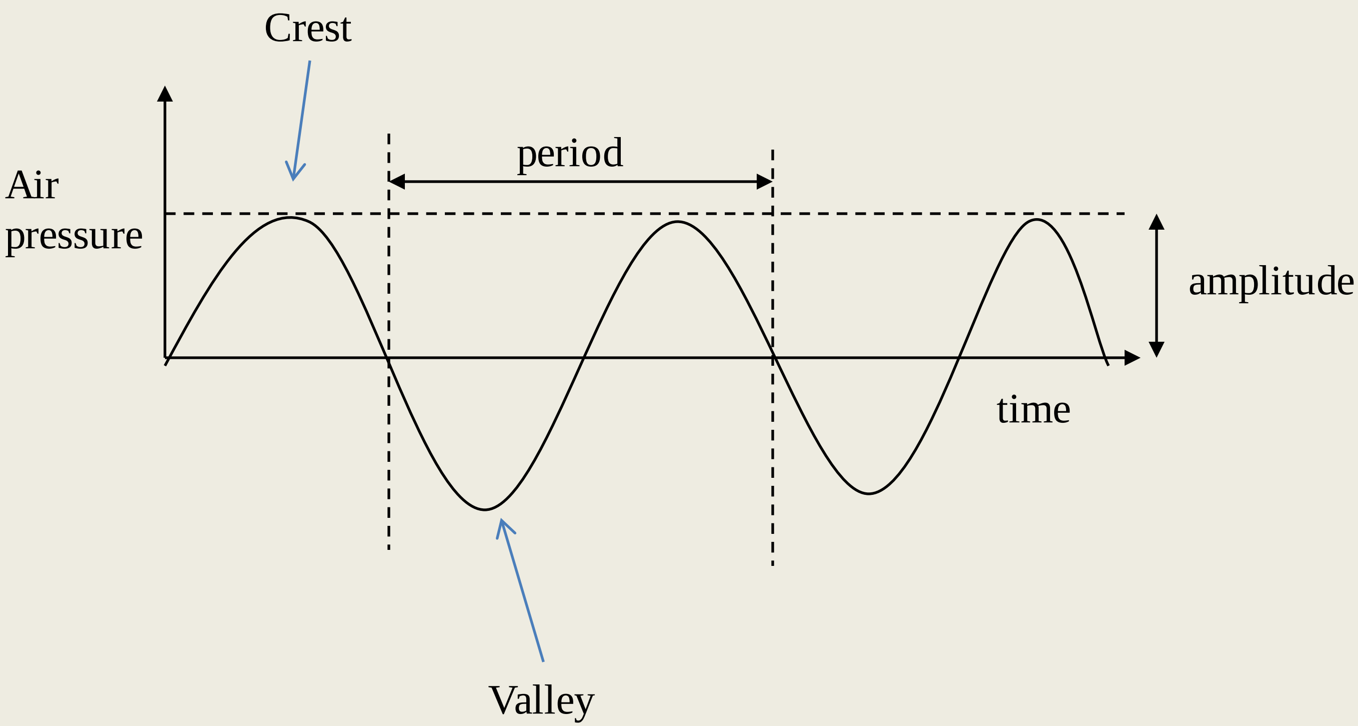
Basic Sound Concepts…

- Waveform repeats the same shape at intervals called a period.
  - **Periodic sound sources** - exhibit more periodicity, more musical - musical instruments, wind etc.
  - **Aperiodic sound sources** - less periodic - unpitched beating, sneeze, cough.

Basic Sound Concepts…

- Sound Transmission
    - Sound is transmitted by molecules bumping into each other.
    - Sound is a continuous wave that travels through air.
  - Sound is detected by measuring the pressure level at a point.
  - **Receiving**
    - Microphone in sound field moves according to the varying pressure exerted on it.
    - Transducer converts energy into a voltage level (i.e. energy of another form - electrical energy)
  - **Sending**
    - Speaker transforms electrical energy into sound waves.

A sound wave



Wavelength is the distance travelled in one cycle

20Hz is 56 feet, 20KHz is 0.7 in.

Basic Sound Concepts

- ***Frequency*** represents the number of periods in a second (measured in hertz, cycles/second).
  - Human hearing frequency range: 20Hz - 20Khz, voice is about 500Hz to 2Khz.

Infrasound	from 0 - 20 Hz
Human range	from 20Hz - 20KHz
Ultrasound	from 20kHz - 1GHz
Hypersound	from 1GHz - 10THz

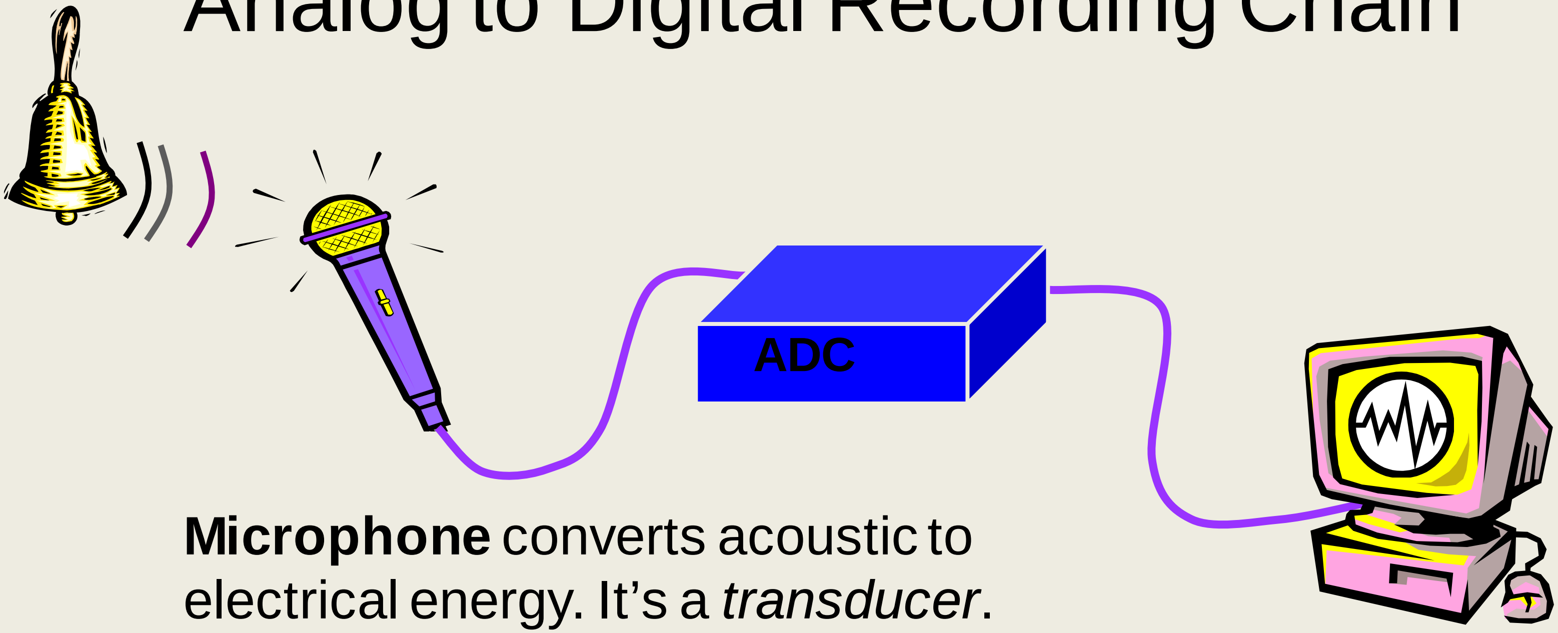
- ***Bandwidth*** –is defined as difference between the highest and lowest frequency contained in a signal

Basic Sound Concepts

- ***Amplitude*** of a sound is the measure of the displacement of the air pressure wave from its mean or inactive state.
  - Subjectively heard as loudness. Measured in decibels.

0 db - essentially no sound heard
35 db - quiet home
70 db - noisy street
120db - discomfort

Analog to Digital Recording Chain



Microphone converts acoustic to electrical energy. It’s a *transducer*.

Continuously varying electrical energy is an **analog** of the sound pressure wave.

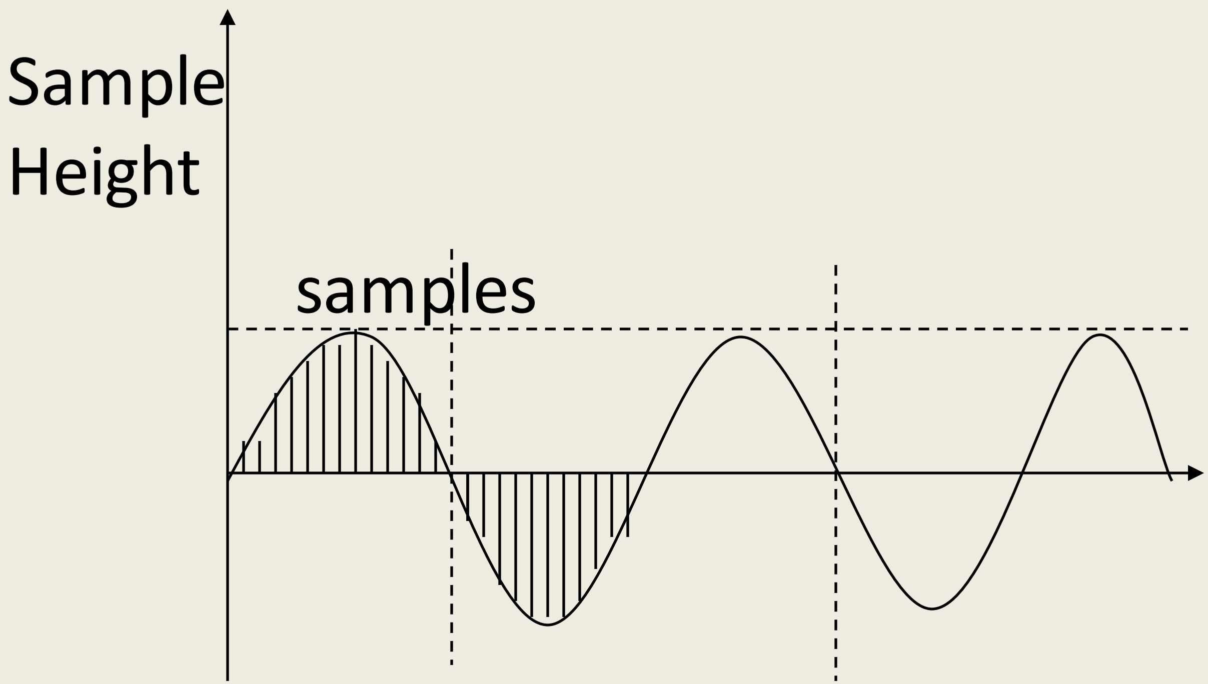
ADC (Analog to Digital Converter) converts analog to digital electrical signal.

Digital signal transmits binary numbers.

DAC (Digital to Analog Converter) converts digital signal in computer to analog for your headphones.

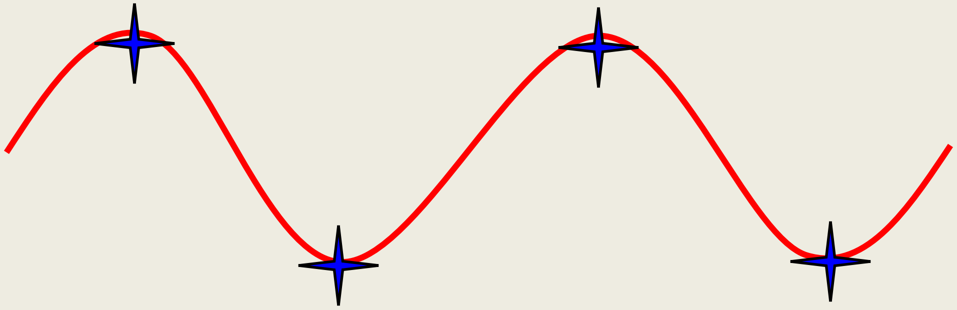
Sampling

Analog signals are converted into digital stream by discrete sampling.



Computer Representation of Audio

- Sampling Rate:
  - rate at which a continuous wave is sampled (measured in Hertz)
    - CD standard - 44100 Hz
    - Telephone quality - 8000 Hz.
  - There is a direct relationship between sampling rate, sound quality and storage space.

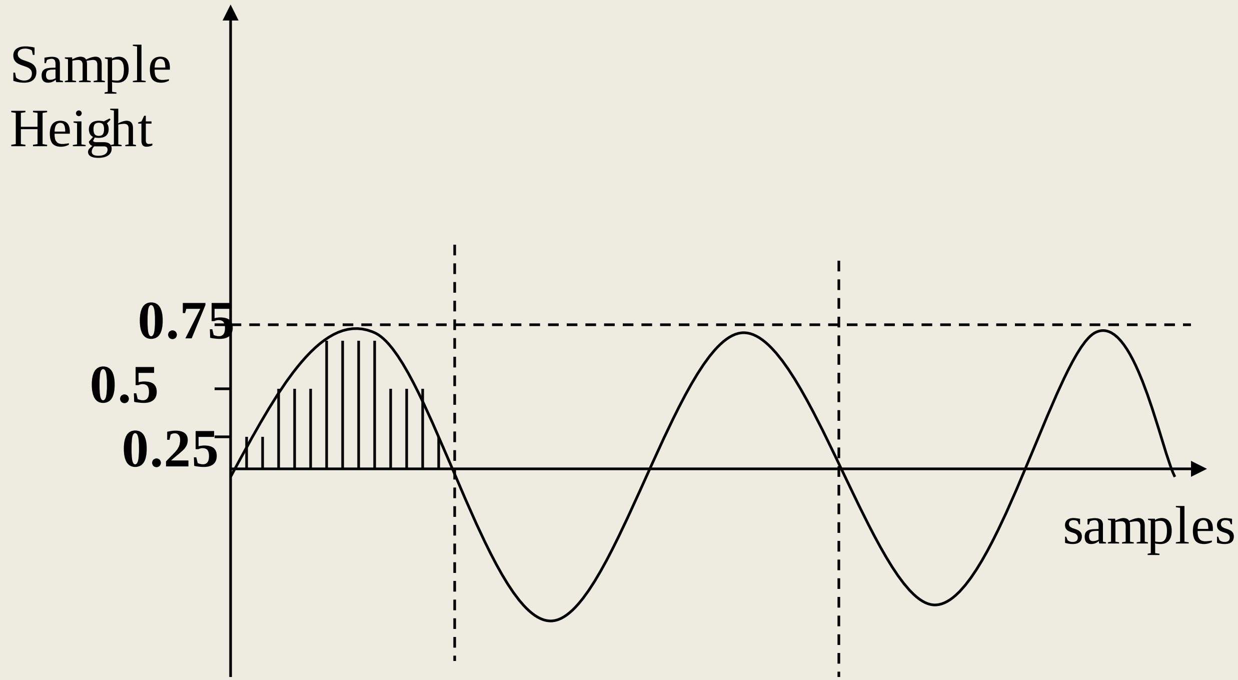


Capturing just the crest and trough(valley) of a sine wave will represent the wave exactly.

Quantization

- **Quantization** depends on the number of bits used measuring the height of the waveform.
- 16 bit CD quality quantization results in 64K values.
- Audio formats are described by sample rate and quantization.
  - Voice quality - 8 bit quantization, 8000 Hz mono(8 Kbytes/sec)
  - 22kHz 8-bit mono (22kBytes/s) and stereo (44Kbytes/sec)
  - CD quality - 16 bit quantization, 44100 Hz linear stereo (196 Kbytes/s)

Quantization and Sampling



Audio Formats

Audio formats are characterized by four parameters

1. **Sample rate: Sampling frequency**
2. **Encoding: audio data representation**
   - μ-law encoding corresponds to CCITT G.711 - standard for voice data in telephone companies in USA, Canada, Japan
   - A-law encoding - used for telephony elsewhere.
   - A-law and μ-law are sampled at 8000 samples/second with precision of 12bits, compressed to 8-bit samples.
   - Linear Pulse Code Modulation(PCM) - uncompressed audio where samples are proportional to audio signal voltage.

Audio Formats

3. **Number of bits used to store audio sample**
   - μ-law and A-law - 8 bit precision, PCM can be stored at various precisions, 16 bit PCM is common.
4. **Channel**: Multiple channels of audio may be interleaved at sample boundaries.

Fidelity

- The closeness of the recorded version to the original sound.
- In the case of digital speech, it depends upon the number of bits per sample and the sampling rate.
- A really high-fidelity (hi-fi) recording takes up a lot of memory space
  - (176.4 Kb for every second of audio of stereo quality sampled at 16 bits, 44.1 kHz per channel).
- Fortunately for most computer multimedia applications, it is not necessary to have very high fidelity sound.

Nyquist Theorem

The sampling frequency determines the limit of audio frequencies that can be reproduced digitally.

- According to Nyquist theorem, a minimum of two samples (per cycle) is necessary to represent a given sound wave.
  - to represent a sound with a frequency of 440 Hz, it is necessary to sample that sound at a minimum rate of 880 samples per second.

Therefore,

Sampling rate = 2 x Highest frequency.

Audio File Size

- Sampling rate:

  44,100 samples per second (44.1 kHz)

- Sample word length:

  16 bits (i.e., 2 bytes) per sample

- Number of channels:

  2 (stereo)

How big is a 5-minute CD-quality sound file?

Audio File Size

How big is a 5-minute CD-quality sound file?

44,100 samples * 2 bytes per sample * 2 channels

= 176,400 bytes per second

5 minutes * 60 seconds per minute

= 300 seconds

300 seconds * 176,400 bytes per second

= 52,920,000 bytes = 50.5 megabytes (MB)

Sound formats and settings

- Recording at high sampling rates produces a more accurate capture of the high-frequency content of the sound.
- Resolution determines the accuracy with which a sound is digitized.
  - The increase in the number of bits in a recording makes the sound playback increasingly realistic.

Sound formats and settings

- Stereo recordings
  - made by recording on two channels, and are lifelike and realistic.
- Mono sounds
  - less realistic, flat, and not as dramatic, but they have a smaller file size.
- Stereo sounds require twice the space as compared to mono recordings.

Sound formats and settings

To calculate the storage space required, the following formula are used:

- ***Mono Recording:***

File size = Sampling rate x duration of recording in seconds x (bits per sample/8) x 1

- ***Stereo Recording:***

File size = Sampling rate x duration of recording in seconds x (bits per sample/8) x 2

Compression

Reducing the physical size of data such that it occupies less storage space and memory.

- Compressed files are easier to transfer
  - amount of reduction in the size of data
  - reduction in the time needed for file transfer
  - reduction in the bandwidth utilization thus providing good sound quality even over a slow network.

Common Compression Methods

- An array of compression techniques have been set by the CCITT Group
  - an international organization that develops communication standards known as "Recommendations" for all digitally controlled forms of communication.
- There are two types of compression:
  - Lossless Compression
  - Lossy Compression

Lossless Compression

Data are not altered or lost in the process of compression or decompression.

- Some commonly used lossless standards:
  - Packbits encoding (run-length encoding)
  - CCITT Group 3 1-D (based on run-length encoding scheme)
  - CCITT Group 3 2-D (based on run-length encoding scheme modified by two-dimensional encoding)
  - CCITT Group 4 (based on two-dimensional compression)
  - Lempel-Ziv and Welch algorithm LZW

Lossy Compression

There is loss of some information when lossy compression is used.

- the object looks more or less like the original.
- This method is used where absolute data accuracy is not essential.
- Lossy compression is the most commonly used compression
- This compression technique is used for image documents, audio, and video objects.

Audio file formats

- m-law: .au, .snd
  - Most frequently used file format on the Internet.
  - not the highest quality audio
  - small in size
  - it has a player on almost all platforms
  - sampled at 8 kHz.
- AIFF(Audio Interchange File Format) .aif , .aiff, .aifc
  - variety of sample rates
  - can be easily converted to other formats

Audio file formats

- Wave: .wav
  - very similar to the AIFF format
  - supports multichannel samples and variety of sampling rates.
  - Follow the RIFF (Resource Interchange File Format)
- MPEG: .mpg , .mp2, .mp3
  - The most popular of the standards today
  - defined by ISO's MPEG (Motion Picture Experts Group)

Audio file formats

- MIDI: .mid, .midi
  - not a specification for sampled audio data
  - a serial communications protocol designed to allow transmission of control data between electronic musical instruments
  - It is a PostScript language for MUSIC

MIDI: Musical Instrument Digital Interface

Use of Audio in Multimedia

- Can use sound in multimedia projects in two ways
  1. **Content Sound**

     provides information to audiences.(dialogs in movies or theater)
  2. **Ambient sound**

     consists of array of background & sound effects

Content Sound used in multimedia

- **Narration** : provides information about an animation that is playing on the screen
- **Testimonials** : these could be auditory or video sound tracks used in presentations or movies
- **Voice-overs** : these are sound for short instructions ,for example, to navigate the multimedia application
- **Music** –music may be used to communicate(as a song)

Ambient Sound used in multimedia

- **Message reinforcement** : the background sound you hear in real life, such as crowd at a ball game, can be used to reinforce the message that you wish to communicate.
- **Back ground music** : set the mood for audience to receive and process information by starting and ending a presentation with music.
- **Sound effect** : sound effects are used in presentation to liven up the mood and add effects to your presentations, such as sound attached to bulleted lists.

Sound editing operations

- Trimming:
  - Remove dead air or blank space from the front and end of a recording
- Splicing and assembly
  - Remove noise from outside that has crept into a a recording
- Volume adjustments
  - When assembling different recordings into a single sound track
  - To provide a consistent volume level

Sound editing operations cont..

- Format conversion
  - Convert the format so that some authoring software can read it
  - Macintosh:AIF, windows: WAV
- Re-sampling or Down-sampling
  - When using lower rates(8 bits) and resolutions than the rate used in recording(16 bits)
  - This process save the disk space

Sound editing operations cont..

- Fade-ins and Fade-outs
  - Useful for sections you wish to fade-in or fade-out gradually
  - This enveloping is used to smooth out the very beginning and very end of a sound file
- Equalization
  - This facility allows to modify a recording's frequency content to sound brighter or darker

Sound editing operations cont..

- Time stretching
  - More advanced programs let you alter the length (in time) of a sound file without changing its pitch
  - This is very useful but must be very careful as it leads to degrade the audio quality severely

Project

- Find all the sound files on your computer: WAV files (.wav), MIDI (.mid or .rmi), and any others (like .mp3 files) that you are able to download from the Web.
- Listen to several files of each format carefully, check their properties (right-clicks properties > advanced), and compare their quality with the properties like the size of file, time of the clip, compression used if any, sampling rate, etc.
- Make a table and record your results in the table.
- Mention whether these sounds are used (or can be used) as content audio or ambient audio, and explain in one sentence the message they give (if any).