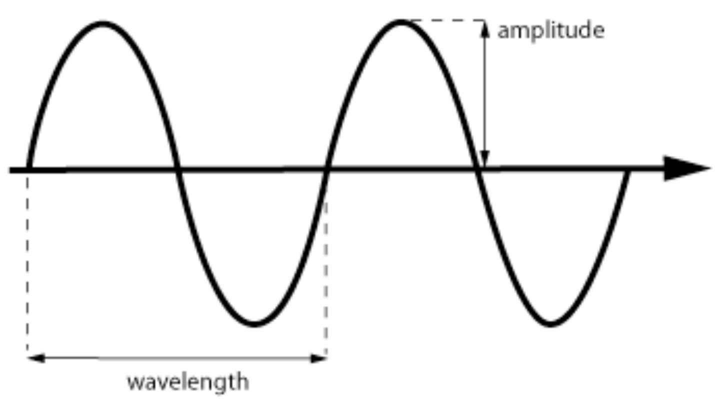
Introduction to Multimedia MMP100
Department of Media Arts and Technology
BMCC CUNY

Overview of Sound

Sound is a **wave** that is generated by an object vibrating in a medium such as air producing changes in pressure. Sound is often represented graphically by a **waveform** that shows changes over time on the horizontal axis and changes in pressure on the vertical axis.

There are a number of terms we use when speaking of the characteristics of sound.

Wavelength is the distance between any point on a wave and the corresponding point on the next wave.



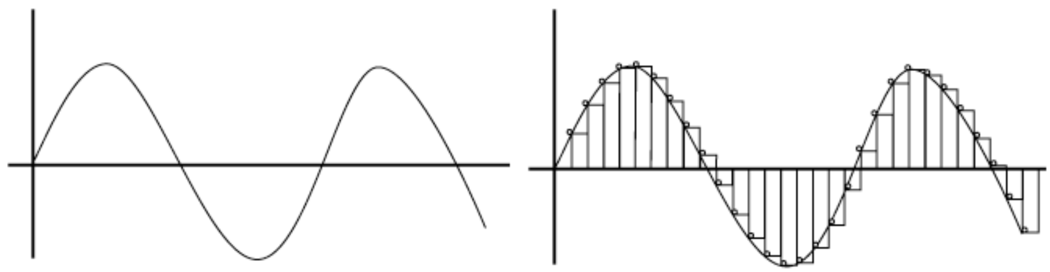
Frequency refers to the number of complete cycles of a wave per unit of time. For example, frequency would be used to measure how many times a sound wave completes a cycle in one second. Frequency is measured in **hertz (Hz)**. The range of human hearing is approximately 20 Hz to 20,000 Hz.
http://science.howstuffworks.com/life/human-biology/hearing1.htm

Amplitude refers to the distance between the highest and lowest point on a waveform. Amplitude determines **volume** or loudness- the higher the peak of a waveform the louder its volume. Amplitude is measured in **decibels (dB)**.

Pitch is related to frequency, the higher the frequency, the higher is the pitch of the sound. If the frequency is lower, so is the pitch of the sound. A sound with a frequency of 20,00Hz would have an extremely high pitch.

Digitizing Sound
Sound is analog- capturing it involves sampling and quantizing.

In the sampling step, the **sound wave** is **sampled** at a **specific rate** into **discrete samples of amplitude** values. If the sampling rate is higher, the data captured will be more accurate. However, a higher sampling rate will generate a larger amount of data, which will require more storage space and processing time. The sampling rate for CD-quality audio is 44100 Hz, or 44100 samples per second.



In the quantizing step, each of the **discrete samples** of amplitude values, obtained from the sampling step, will be **mapped** and rounded to the nearest value on a scale of discrete levels (bit depth). The more levels available in the scale (the higher the bit depth), the higher the accuracy in reproducing the sound—referred to **as higher resolution**. However, higher resolution will require more storage space.

Sampling Rate
This refers to how many times per second the sound is sampled. Harry Nyquist and Claude Shannon figured out that to be able to accurately record and reproduce a wave, sound wave in our case, you have to sample at a little more than twice the highest frequency.
http://en.wikipedia.org/wiki/Nyquist%E2%80%93Shannon_sampling_theorem

This is important when you want to digitize audio. We know that humans can hear up to 20,000Hz so that means we need to sample at over 40,000 times per second to accurately record the sound. This is why CD audio uses 44,100 samples per second.

Bit Depth
The second part of digitizing audio is turning the sample taken into a number. This is the quantizing part. In order to create that number we need a scale to measure the audio. Since everything in the computer is binary, that scale will have a binary number. If you think of the scale like a ruler, then the bit depth determines how many marks there are

on the ruler. The more bits, the more marks, the more marks the more accurate the number is that we are recording.
Each time you add a bit you double the number of marks on the ruler. For example CD audio is recorded at 16bits per sample. This means that there are 65,536 marks on the ruler. If you move up to use 24bits per sample (50% more bits) that means there are 16,777,216 marks on the ruler (25,600% more possible numbers).

Stereo vs Mono vs 5.1 vs 7.1 and beyond
When you save digital audio you have to use at least one channel of sound. This is Mono. When you have two channels of sound you can have different sounds coming out of the left and right headphones or speakers. That is Stereo sound. 5.1 and 7.1 audio have 5 and 7 channels of audio each plus a low frequency (bass) channel: the .1 part. Most music you get off of iTunes is Stereo or Mono while audio from a DVD or Blu-ray is usually 5.1 or 7.1. Formats with even more channels exist.

Data Rate
In general data rate is measured in bits per second. For Sound, data rate is how much data is needed per second to play a sound. This is important especially for streaming sound because you want a data rate that is high enough to make the sound accurate (fidelity) and yet low enough to be able to stream. You can calculate the data rate by multiplying the Sampling Rate * Bit Depth * Channels
CD audio for example would be
44,100 samples/second * 16bits/second * 2 channels = 1,411,200 bits per second
This is usually shown in kilobits so 1,411kbps

Compression
The data rate for your home internet connection is usually measured in megabits (Mbps). While many cable and fiber-based home connections are now 10+Mbps that wasn't always the case. CD audio in megabits is 1.411Mbps. Because most people couldn't play that a way of compressing the sound to make is smaller was needed.

In stepped MP3. It is a compression algorithm and file format that cuts down the data rate by 10 times or more. There are other formats that do similar compression, like the AAC format Apple uses in iTunes. iTunes songs were originally released at 128kbps (more than 10 times less the 1,411kbps of CD). As our internet speeds and storage space has grown people now use higher bit-rate audio files. Apple now allow 248kbps songs in iTunes.

Problems with Compression
MP3 like jpeg is a lossy compression scheme. This means data is lost when converting a song to MP3. If you set the data rate too low then you can tell the difference and hear that sounds are missing from the original recording

Links on Digitizing Audio

http://entertainment.howstuffworks.com/analog-digital2.htm
http://www.apple.com/itunes/mastered-for-itunes/docs/mastered_for_itunes.pdf

Some Common Digital Audio File Formats

File Type	Abbreviated From	Compression and Other Information	Created by
.aiff	Audio Interchange File Format	Usually uncompressed, though there is a compressed version	Apple Computer with Electronic Arts
.wav	Waveform Audio File Format	Uncompressed and also a number of compressed versions	Microsoft and IBM
.mp3	MPEG-2 Audio Layer 3	Highly compressed; common audio format for consumer audio storage	Moving Picture Experts Group
.aac	Advanced Audio Coding	Highly compressed. Designed to replace .mp3 format, higher quality sound with similar bit rates	Group of companies including AT&T Bell Laboratories, Fraunhofer IIS, Dolby, Sony Corporation and Nokia

MIDI

The MIDI format is another way of storing music information. MIDI (Musical Instrument Digital Interface) is a communications protocol that defines a common interface for electronic digital music instruments to communicate with computers, or other instruments or devices containing microprocessors. It specifies the configurations of cables and cable plugs as well as the format of the data. MIDI is a way of describing information and a set of agreed upon standards, not based on an audio waveform.