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Feb. 10, 2023

The Digitization of Music

Have you ever wondered how exactly music gets recorded in a studio and then played back on our mobile devices? Sound playback technology dates back to Thomas Edison's phonograph in 1877 (Brain 2000). This was a device that recorded sound by using soundwaves to vibrate a diaphragm that moved a needle, physically etching the recording into tinfoil. This is what we could call an analog recording device, since the soundwave recording was a continuous representation of the soundwave. Another analog playback device you might be familiar with is the record player, where a needle passes over the grooves of a disk which gets converted into an electrical signal and then amplified into sound. Nowadays, the vast majority of music we listen to is stored digitally, for example on Spotify or Apple Music. This allows us to have vast catalogs of music just a tap away on our smart devices. This begs the question, how is sound stored digitally and played back essentially perfectly recreated?

What exactly is sound? Sound is a physical wave that can contain many different frequencies.

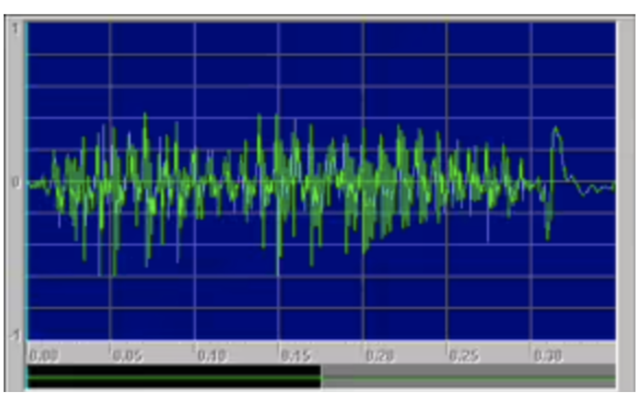

Fig1. Sound wave in time domain for the word "hello", containing many frequencies (Brain 2000)

For a pure tone, we would expect to see just a single sine wave of that frequency. For speech, the wave would look more complex because the sound wave would consist of multiple frequency components. Since sound is an analog wave that computers cannot understand, an ADC (analog-to-digital converter) is used to record the sound. The most important parameter of this conversion process is the sampling rate. The sampling rate tells us how many times per second the analog signal is being sampled, or measured.

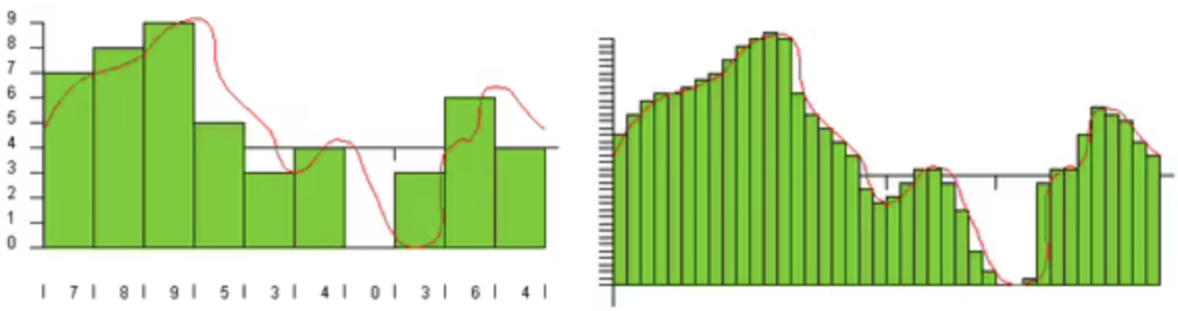

Fig2. Left side signal sampled at a low rate; right side signal sampled at a high rate (Brain 2000)

Using too low of a sampling rate would result in a poorly digitized sound wave that loses a lot of information from the original analog signal. As a result, the sound would not be able to be reconstructed accurately and played back. Sampling below the Nyquist Rate, twice the frequency bandwidth, would result in distortion called aliasing where signals cannot be distinguished. In general higher sampling rates are better to get the best recreation of the analog signal and the highest fidelity sound. Today audio is generally recorded at a sampling rate of 44.1kHz (TechGuru 2020). So music is recorded in a studio, and that recording is fed through an ADC so that it can be stored as digital information that a computer can read.

Once that digital information is stored, how is it played back so that we can listen to the music as it originally sounded in the studio? That digital information has to be converted back into an analog signal, aka sound that we can hear. To do this a circuit called a DAC (digital-to-analog converter) is used. DACs can be found in your playback device, so your smartphones and laptops. There are several parameters that can set apart a good DAC from a great DAC, which is why many audiophiles prefer to use external DACs to get the best music playback possible. One parameter that limits dynamic range is bit depth. Bit depth refers to the amount of information in each sample, which limits the volume levels that notes can take on (Thomas 2022). A low bit depth would result in the notes not being able to be recreated how they were intended, with the musical color given to them through changes in note emphasis. Higher bit depth is better, but generally after 16 bit a higher bit depth is unnecessary to the average listener. With a higher bit depth, however, comes the requirement for the DAC to convert more information at once, so it will need a high bit rate to keep up. The bit rate is how fast the music data is being decoded by the DAC (Thomas 2022). There are different types of files such as the MP3 which has 320kbps and the FLAC which has 1400+ kbps. The higher the bit rate the higher fidelity the sound, but the more space the file will take up in storage. Like all of these parameters, a middle ground is probably best for most listeners, so the MP3 would work just fine. Once the DAC has converted the digital signal back into an analog signal, an amplifier amplifies the signal so that you can listen to your music through the speakers.

So what makes a good listening experience? You will need to have a good sampling rate for your music recording so that no information from the original signal is lost. Then it would be best to have a high bit depth with 16 bit being the typical value. Lastly if you are choosing space efficient compressed audio over lossless then you need to look for a high enough bitrate so that the sound's fidelity is not diminished. While many still enjoy analog storage forms for listening to or collecting music, digitization has allowed us to have convenient access to music wherever we are.

Bibliography

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