**Lecture 1 – Sound: an overview**

A sound wave traveling through air has a root mean square pressure, which correlates to its loudness. For the human ear, a logarithmic conversion from the root mean square pressure (unit of pascal) is used to better correlate with precepted loudness. The simplest type of wave is called a sinusoidal wave. This periodic wave has many characteristics that define it: Wavelength [m], period [s], amplitude, frequency [Hz] and phase [rads]. All waves, including sound waves, can be represented with combinations of these sinusoidal waves. The Fourier transform is used to find the sinusoids that compose a wave.

In the modern world, most sound is processed digitally with computers. To do this, the analog wave must be digitized via an A/D conversion. The Fourier transform adapted to computers, the Fast Fourier Transform, is used for this conversion. When converting waves into a digital format, many things must be considered, such as the sampling rate. Sampling rate refers to the interval that the sound wave is sampled to generate a digital format. This must be at least twice the highest frequency of the sample, to not introduce aliasing into the result. During the A/D conversion, the signal levels of the sample are quantized, which means assigning a set value to each sampled point. The more values in the set, the higher the bit depth of the resulting digital audio is. For a real-life example, MP3 with higher bit depths have higher fidelity than ones with lover bit depth.

**Lecture 2 – Acoustics**

Loudness is different from the sound pressure. They are correlated, but loudness relates to the sound level perceived by humans. We found out different values in the loudness dB scale that correspond to different perceptions, such as hearing thresholds, examples of loudness and discomfort limits. As for the sound pressure for multiple sound sources, the signals are summed like the superposition of waves. For example, if two sources play the exact same sound, the RMS is increased by 3 dB. The distance from an audio source spreads spherically, so as distance from the source increases, the RMS is inversely proportional to distance. This means that the RMS level drops by 6 dB every time the distance doubles.

Overall, the lecture contents included good introductory concepts related to the course. I found the examples given very helpful. Although most of the things introduced in the beginning lectures were taught in high school or the basic physics courses, they were good refreshers.