**AUDIO ACQUISITION**

**RECORDING**

Sound recording is an electrical or mechanical inscription and re-creation of sound waves, such as spoken voice, singing, instrumental music, or sound effects. The two main classes of sound recording technology are **analog recording** and **digital recording**. Acoustic analog recording is achieved by a small microphone diaphragm that can detect changes in atmospheric pressure (acoustic sound waves) and record them as a graphic representation of the sound waves on a medium such as a phonograph. In magnetic tape recording, the sound waves vibrate the microphone diaphragm and are converted into a varying electric current, which is then converted to a varying magnetic field by an electromagnet, which makes a representation of the sound as magnetized areas on a plastic tape with a magnetic coating on it.

The purpose of recording is to capture the best possible signal. Regardless of the recording situation, one should consider at least three important acoustic parameters: frequency response, dynamic range, and signal-to-noise ratio(SNR).

**Frequency response** describes the range of frequencies captured in the recording, measured in Hertz (Hz).

**Dynamic range** is the ratio of the loudest to the softest part of the signal, measured in Decibels (dB).

**SNR** is the ratio of signal amplitude to the amplitude of noise usually generated by the circuit, measured in Decibels (dB). In the audio digitization process, signal-to-noise ratio refers to the ratio of the maximum signal power to the quantization noise generated by the analog-to-digital converter.

Digital recording and reproduction converts the analog sound signal picked up by the microphone to a digital form by the process of digitization. This lets the audio data be stored and transmitted by a wider variety of media. Digital recording stores audio as a series of binary numbers representing samples of the amplitude of the audio signal at equal time intervals, at a sample rate high enough to convey all sounds capable of being heard. Digital recordings are considered higher quality than analog recordings not necessarily because they have higher fidelity, but because the digital format can prevent much loss of quality found in analog recording due to noise and electromagnetic interference in playback, and mechanical deterioration or damage to the storage medium.

**QUANTIZATION**

Quantization is the process of mapping a large set of input values to a (countable) smaller set, such as rounding values to some unit of precision. A device or algorithmic function that performs quantization is called a quantizer. The round-off error introduced by quantization is referred to as quantization error.

**SAMPLING**

In signal processing, sampling is the reduction of a continuous signal to a discrete signal. A common example is the conversion of a sound wave (a continuous signal) to a sequence of samples (a discrete-time signal). When it is necessary to capture audio covering the entire 20–20,000 Hz range of human hearing, such as when recording music or many types of acoustic events, audio waveforms are typically sampled at 44.1 kHz, 48 kHz, 88.2 kHz, or 96 kHz. The approximately double-rate requirement is a consequence of the Nyquist theorem.

**DIGITIZATION**

Digitization is the representation of an object, image, sound, document or a signal (usually an analog signal) by a discrete set of its points or samples. The result is called digital representation or, more specifically, a digital image, for the object, and digital form, for the signal.

**Analog-to-Digital conversion and digital data transfer**

In order to perform acoustic analysis on recorded speech data, the audio signal has to be converted into a digital audio file format, such as Wav or Aiff. Analog recordings have to be digitized and digital recordings need to be transferred to a personal computer via a digital audio file transfer interface. This is an important, yet often underestimated, stage in the process of preparing audio data for analysis. The main goal of A/D conversion (digitization) is to obtain the best possible digital representation of the original analog waveform. One should choose a sample rate that will capture a broad range of frequencies and a bit-depth that will allow a wide dynamic range and a negligible amount of quantization noise. These goals can be achieved by means of a premium-quality, stand-alone A/D converter operating at the sample rate of at least 48,000 Hz and a 24-bit resolution.

**Python libraries to be used (tentatively)**

**PyAudio** - Python bindings for [PortAudio](https://wiki.python.org/moin/PortAudio) audio input and output

**Wave** - Provide an interface to the WAV sound format

**Android Applications for Audio Acquisition**

* Easy Voice Recorder
* RecForge Pro
* Smart Voice Recorder