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# Abstract

Sampling is a process in which one takes a continuous time signal (analog signal) and transforms it to a discrete time signal (analog to digital conversion (ADC)), by taking samples for a specified amount of time over a period (sampling period). After sampling is complete one can then apply quantization to encode the discrete values in a 0 and 1 format, which allows a computer to process a signal. After quantization, the noise can be removed from the signal by using filters, the signal to noise ratio can be calculated, and the original signal can be optimized and recovered.

# Introduction

One of the major reasons from sampling a continuous time signal is for a computer system to be able to modify its properties. After this is complete original continuous signal can be reconstructed from the samples and collected by a receiver. In our case the receiver will be someone listening to the sound of a music file that has been modified or filtered to remove unwanted noise. In order to accurately reconstruct the analog signal, we must sample along the sinusoidal signal at the appropriate points. If sample to much could waste processing time and if sample too little one may not be able to accurately recreate the continuous signal. In order to reconstruct the signal it is best to take the highest frequency in the signal and multiply this by two to get the number of time per period (T) that we must take a sample. The basic function of a quantizer is to electronically define a range of input values, subdivide that range into a set of subregions, and then decide within which subregion the input sample lies. The next step is coding which generates the binary word corresponding to the assigned level [AppliedDSP]. In our project the input of the wav file will be encoded with a particular algorithm that will compress the very large signal into a much smaller signal by representing certain subsections of the signal with binary digits (many-to-one mapping). When analog to digital conversion happens there will be a loss of data or error. [AppliedDSP].

## DSP Algorithm Design

***The key questions related to quantization are***

1. Into how many intervals *N* can we split the original interval, or stated in terms of binary representations, how many bits *B* can be used, where N = 2^B

2. Given that *x*[*n*] falls into *Qk*, how should *x’*[*n*] be chosen.

3. Given the set of approximate values *x’*[*n*], how can the continuous-time approximation *x*(*t*) be reconstructed.