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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.