Audio Spectrum Analyzer and Manipulator  
 Software Design Document

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# Introduction

## Purpose

This software design document describes the architecture and system design of an audio spectrum analyzer that is also capable of editing the audio data.

## Scope

This software aims to analyze the frequency spectrum of an audio file. The main goal is to develop an intuitive program for displaying the applications of audio manipulation through use of Fourier analysis.

## Definitions and Acronyms

ASA: Audio Spectrum Analyzer

FFT: Fast Fourier Transformation

LV: LabView 2014 32-bit

# System Overview

The basis of this software is systematic manipulation of the input audio file in the frequency domain. Transition between the time domain and frequency domain is accomplished through the application of Fourier Transformation. Post transform the magnitudes of each frequency within the spectrum are available and can be plotted and manipulated. A reverse Fourier transformation can be performed to return the file to a time domain data stream which can be played. When computationally analyzed, a discrete (rather than continuous) Fourier transform is used, specifically the fast Fourier transform (FFT) which factors the discrete Fourier matrix into a product of predominantly zero factors.

# System Architecture

## Architectural Design

The software begins by reading in the audio file from a specified directory, noting the format must be in one of the approved formats (.wav, .mp3, .mp4). Once read in, a FFT will be used to obtain the frequency spectrum of the audio file. A three dimensional plot of the frequency spectrum will be generated for the entirety of the audio file in sampling of one-second clips. Using these frequency spectrums the audio data will be filtered using both a low-pass and a high-pass filter, with user-specified cut off frequencies, to generate two additional spectrum plots post filtering. Options to play the original audio file, the low-passed audio file, and high-passed audio file will be available to compare the effects of filters on the timbre.

## Design Rationale

Three dimensional plots allow the user to visually observe the frequency spectrum of the audio file and observe the application of filtering of frequencies. The audio play options will add an auditory dimension so as to best observe the effect of filtering on the timbre of the audio file.

FFTs are used to efficiently manipulate the audio file data in order to produce a the visualization and analysis in a time-efficient manner.

# Human Interface Design

## Overview of User Interface

Upon running the VI, the user will be prompted to designate the directory path of the audio file. This will automate the analysis of the audio file and produce the three 3D plots described above. Additionally there will be push buttons to play the various versions of the audio files. Manipulation of cut-off frequencies will be possible through user designated input as well

## Screen Images