Reducing Audio Bandwidth with an FFT-based Low-Pass Filter

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***Abstract*—The abstract goes here.**

I. Introduction

Filtering an audio signal is extremely important for many reasons. When an appropriate filter is designed and used to remove any unwanted frequencies, then the bandwidth and power necessary for audio signal transmission is reduced. Why is this important? The frequency bands used for the transmission of many types of signals are scarce resources. Every transmitter that can interfere with others has to operate in a licensed band and is subject to bandwidth limitations. This is why multiple radio stations can operate in the same geographical area. They are assigned different center frequencies and they have a certain bandwidth they can occupy. If one radio station transmits beyond their assigned bandwidth, they will impact the signal transmission of other local radio stations. One way to ensure that a radio station stays within their assigned bandwidth is through the use of filtering. [2]

Additionally, if you travel to a different city, you will find radio stations that operate at the same frequency as the radio stations located locally here in Atlanta. This is possible because radio transmitters are limited in power. With unlimited power, for example, one could hear FM 106.7 throughout the United States and no other radio station would be able to use the frequency 106.7 MHz. But power is limited, and high-power radio transmissions cost a lot of money [2]. This is why some radio stations like to brag about how powerful their transmissions are. It is also why start-up radio stations have a very limited range. Therefore, if power costs money, one does not want to waste it by boadcasting frequencies that most humans cannot hear.

Both bandwidth and power savings are the motivation behind Fast Fourier Transform (FFT)-based low pass filters. This paper will demonstrate how a Finite Impulse Response (FIR) low pass filter can remove unecessary frequencies in order to reduce the bandwidth and power required for transmission.

A. Objective

Our objective is to design and demonstrate an FFT-based low-pass filter to reduce bandwidth of a recorded audio signal for the purposes of AM Radio transmission.

II. Methods

We collected data, designed a low pass filter, pushed the signal through the filter, and then evaluated the resulting signal.

A. Data

We collected an audio signal by recording the song 'Flight of the Bumblebee' at the full frequency spectrum that a compact disc (CD) is recorded at, which is 44.1 kHz. The human ear does not even hear this full spectrum. This means that there are potential bandwidth 'savings' for transmission! This collected audio recording is referred to as the 'raw signal.'

B. Filter Design

The typical Human can only hear frequencies in the range of 20Hz through 20kHz, and this range only decreases as humans age [3]. Using the website http://onlinetonegenerator.com/hearingtest.html, we will test the hearing range of each member listening to our final presentation. This demonstration will provide a tangible example why the appropriate filtering of an audio signal will have little to no impact on the quality of sound that a person may hear after the filtering.

When our project group performed the hearing test, the highest audible frequency was 15kHz. Since our project group was not able to hear the frequencies above the 15kHz threshold, any low pass filter that removed the high frequencies above 15kHz would have no impact on what we hear. Removing the unecessary frequencies from an audio signal transmission provides a few nice benefits such as the following:

1) It reduces the necessary power for transmission of the audio signal
2) It reduces the necessary bandwidth for transmission

Given the human threshold for hearing, an obvious opportunity is to apply low-pass filtering in order to transmit audio signals with smaller bandwidths. The ideal low pass filter would completely eliminate all frequencies above a certain cutoff point (in our hearing test example, the cutoff would be at 15kHz) while passing all frequencies below the cutoff point [5].

In order to filter out higher frequencies in an audio signal transmission, we will build a low pass filter. Specifically, we will describe how a low pass filter is created using a finite number of non-zero filter coefficients which is called a *Finite Impulse Response* filter or FIR. Given the impulse response, we can find the coefficients of the filter, and vice versa [2]. For example, Figure 1 shows a graphical depiction of a low pass filter that begins to cut out the frequencies above 1 rad/s. Frequencies below the cutoff frequency are considered to be in the “Passband,” or allowable frequency band. Frequencies above the cutoff frequency filtered out and are considered in the “Stopband.”

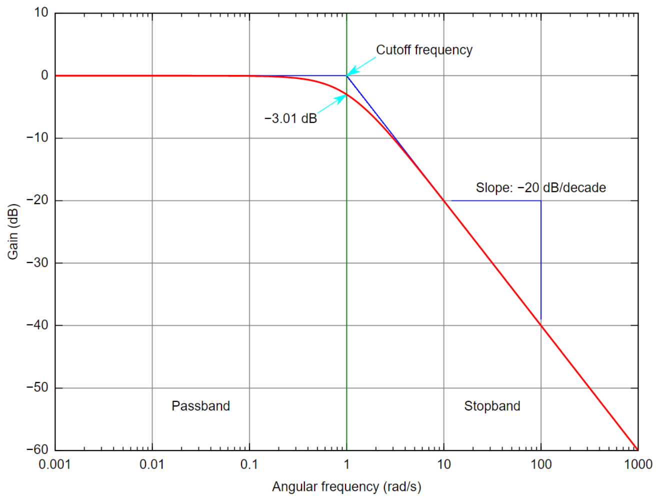

Fig. 1. Example of Low Pass Filter: https://upload.wikimedia.org/wikipedia/commons/6/60/Butterworth_response.svg

C. *Nyquist-Shannon Sampling Theorem*

When recording audio signals, one has to set a sampling frequency (or a frequency for recording the signal). There are several reasons why we set a sampling frequency:

1) Data storage conservation,
2) Bandwith conservation,
3) Power conservation.

III. Evaluation of Filtered Signal

We reduced the raw signal from 44.1 kHz to () kHz, with no perceived quality degradation to the three judges.

IV. Conclusion

The conclusion goes here.

References

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