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

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

I. Introduction

The transmission of audio signals through radio frequency is bandwidth-limited, which is why the FCC regulates transmissions so closely. Keeping the frequency higher saves bandwidth, and the goal is to trim the frequencies without impacting the quality of what the human ear perceives.

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 the presentation. When the hearing test was performed within our group, the highest audible range was 15kHz. With this simple demonstration, the audience will understand why a low pass filter that filters frequencies above 20KHz would not impact the quality of the sound. Understanding that humans cannot hear above and below certain thresholds creates an opportunity transmit audio signals with smaller bandwidths.

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,

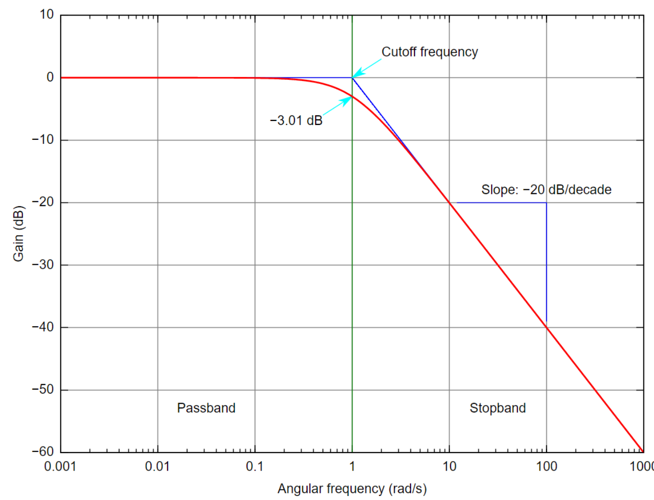

Fig. 1. Example of Low Pass Filter: https://upload.wikimedia.org/wikipedia/commons/6/60/Butterworth_response.svg

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

[1] *GNU Radio*, https://www.gnuradio.org/.
[2] Baxley, R., Henshaw, A., Nowlan, S., & Trewhitt, E. *Software-Defined Radio with GNU Radio: Theory and Application, Georgia Tech Professional Education course notes.* (2017)
[3] Rossing, Thomas (2007). Springer Handbook of Acoustics. Springer. pp. 747, 748. ISBN 978-0387304465.