‘This case study has no filter’

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Abstract - In this case study, the basics of sound engineering in the context of signal modification were used to filter out unwanted noise. By using concepts from electric circuits and signals systems one can use MATLAB so simulate an equalizer that will filter out unwanted signals. Several sound signals were analyzed to remove undesired frequency components as well as giving the listener control of boosting or minimizing different frequency bands of the signal. After altering the input sound signal to the listener’s specifications, the frequency response was evaluated defining the system in more compact terms. Lastly the relationship between the frequency and time representations of the signals and alterations were explored and shown to support the mathematical foundation of signal processing.

Background

Equalizers are in every modern car, where a person can tune their radio with bass, treble and unity. In this case study, we were tasked to create an Equalizer using our knowledge of electric circuits and signals and systems. The processes that modify and enhance audio signals occur through filters that manipulate the components and frequencies of an input signal. Physically, these filters can be built and modeled as circuits using wires to carry the signal throughout the filter as it gets processed, and circuit elements to perform different tasks within the filter. Mathematical relationships can be found for these filters via circuit analysis.

The objective of this study is to delve into the underlying mathematical systems at play to fine tune the sound signal to the exact specifications of the listener. By gaining these skills, several distorted audio signals can be made clearer through frequency analysis and mathematical relationships such as the impulse and frequency responses can be leveraged to make the system simpler and more efficient.

This case study also tests our understanding of other mathematical concepts like the Fourier transform to convert from time domain to frequency domain, as well as the use of convolution. Other key ideas that were used were complex exponentials being used as an input to a low or high pass system, which allows us to see the full extent of the frequency response for different frequency inputs. Other important relationships that can be seen through this cause study are how tau = R*C the time constant determines the behaviors of the impulse response. All these concepts were used in the designing of a full equalizer.

Formulas

$f_c = \frac{1}{2\pi RC}$ [1]

This formula is known as the cutoff frequency

$$x[k] = \sum_{n=0}^{N-1} x[n] e^{\frac{-j2\pi kn}{N}}$$

[2]

Fourier transform

$$y[n] = \sum_{k=-\infty}^{\infty} x[k] h[n-k]$$

[3]

convolution

Methods

A. Creating a full equalizer

We designed a full equalizer with the following audio presets/ equalizer settings: bass, treble, and unity. Our system was designed with 6 frequency bands. Each system contained two frequency bands. Bass boost targets lower frequencies, so the most straightforward design for this preset was a filter that would attenuate frequencies above a certain point in an arbitrary signal. By attenuating the high frequencies and allowing the low frequencies to pass, this filter produces the boost's desired outcome of targeting an increase in low frequencies relative to high frequencies. A low-pass filter can be created as a circuit by placing a resistor in series with a capacitor that lies across the output terminal of the system as in Figure 1.

For this equalizer setting (bass), it was decided to use a low pass in order to obtain lower cutoff frequencies. 2 individual bands were created in this setting using low pass filters. The lowest band was 60 hertz and the other was 230 hertz. We were able to create a circuit whose cutoff frequency was 60 and 230 hertz. Using the cutoff frequency formula and knowing the desired frequencies we let the capacitor be 1e-6 and found the resistor necessary to create that cutoff frequency.

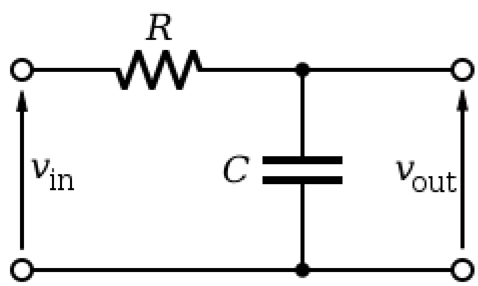


Figure 1: A circuit configured so that the resistor (R) is in series with a capacitor (C) that lies across the output terminal at V_{out} to act as a filter (RC circuit).

The 1st band was a 60 hertz low pass, the second band was 230 hertz. Both signals were then convoluted with the original signal to create a linear combination containing both bands. This was the approach to creating the bass equalizer settings.

The unity preset was designed to approximate a flat frequency response from an arbitrary signal. The most straightforward design for this preset was via the use of a filter that would attenuate frequencies both above and below certain points in an arbitrary signal. By cascading or combining the low-pass and high-pass filters, such an effect can be achieved. In doing so, a bandpass filter is effectively created, and is configured as in Figure 3.

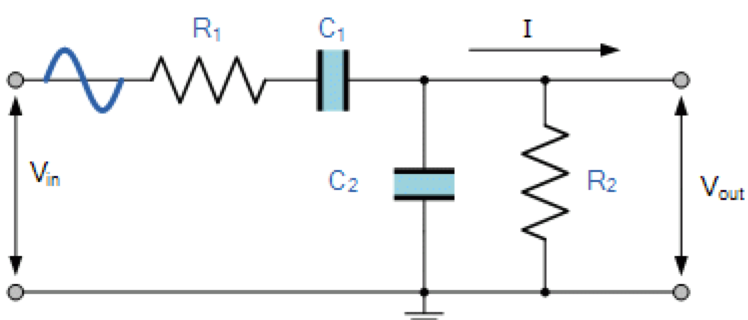


Figure 2: A circuit configured so that low-pass filter is in series with high-pass filter that lies across the output terminal at V_{out}, also called a bandpass filter.

The first loop can be identified to match Figure 2 which represents a low-pass filter, and the second loop can be identified to match Figure 3 which represents a high-pass filter. When a signal is passed through, the first loop attenuates any frequencies above a certain point allowing the frequencies below that cutoff to pass, and then the following loop attenuates the remaining frequencies below a certain point, ultimately allowing the frequencies between the two cutoffs to pass through. For this reason, I set the cutoff frequency for the high pass to 910 hertz and the low pass 230 hertz. This also defined the bandwidth of the bandpass by subtracting the top from the high f_h − f_l. The same process was used to create the bands, where convolution was used to create a linear combination. In addition, a length of about 1000 points was used to see the full mathematical relationship.

Treble boost targets higher frequencies, so the design for this preset was a high pass filter that would attenuate the frequencies below a certain point. By attenuating the low-frequency signals and allowing the high-frequency signals to pass, this filter would be able to achieve the treble outcome of targeting an increase in high frequencies relative to low frequencies. A high-pass filter can be created as a circuit by placing a capacitor in series with a resistor that lies across the output terminal of the system as in Figure 1 in which the output signal, V_{out} is equivalent to the voltage across the resistor, V_R.

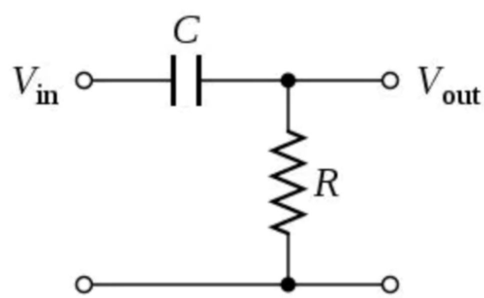


Figure 3: A circuit configured so that the capacitor (C) is in series with a resistor (R) that lies across the output terminal at V_{out} to act as a filter (CR circuit)

The full equalizer was created with a vector containing the linear combination of the different bands (totaling to 6 band frequencies). To adjust the setting, the bands were multiplied by coefficients that represented gains in db. The gains in dB went up by 1 when the coefficients were increments of 10 and down when increments of .10. This is critical as the wrong adjustment of the gain setting and the noise can sound static. This is vital information that can be used to filter out noises in signals.

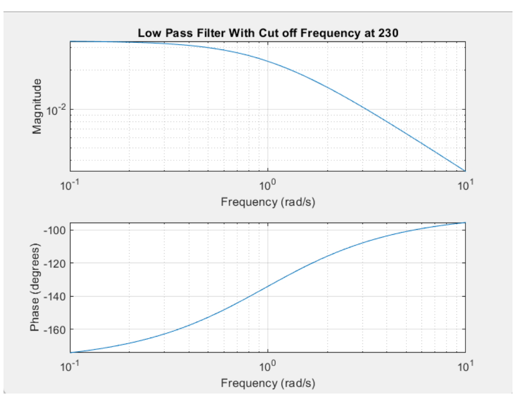
B. Filtering out noise

Filtering out noise can be done using the above information. In order to filter out the noise it is important to locate the location of the frequency at which the noise is coming from. The first step I took was running the sound through a spectrograph in order to see at what frequency was the noise coming from. It was important to see where the noise was coming from to fluctuate our coefficients/gains (in dB). In other words, when I ran the sound, I found noise at 60 hertz so I would decrease near the noise and raise the other frequencies. The second step was adjusting the equalizer to obtain the original signal.

When analyzing other arbitrary signals, the same process was used to see where the frequency of the sound was. I then fixed the equalizer setting to match the signal and tried to improve the signal.

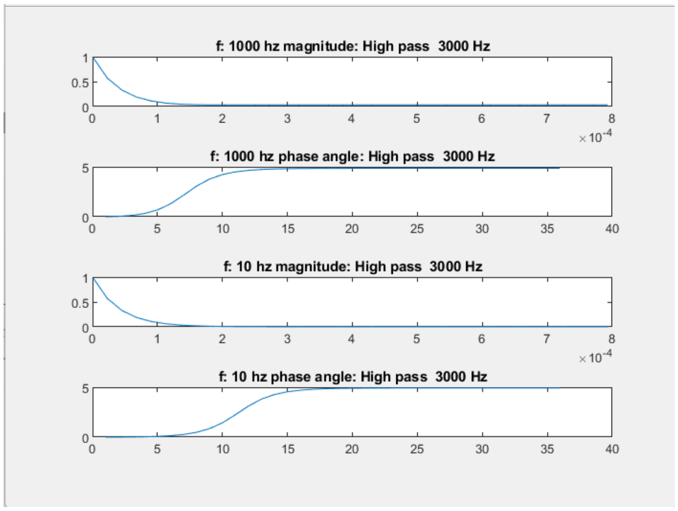
Results

When constructing the full equalizer, it was important to check the frequency response of the bands and the setting (treble, bass, or unity) in order to see whether it is working properly. The first figure shows the low pass with a 230-cutoff frequency, the lowpass did as expected behavior.

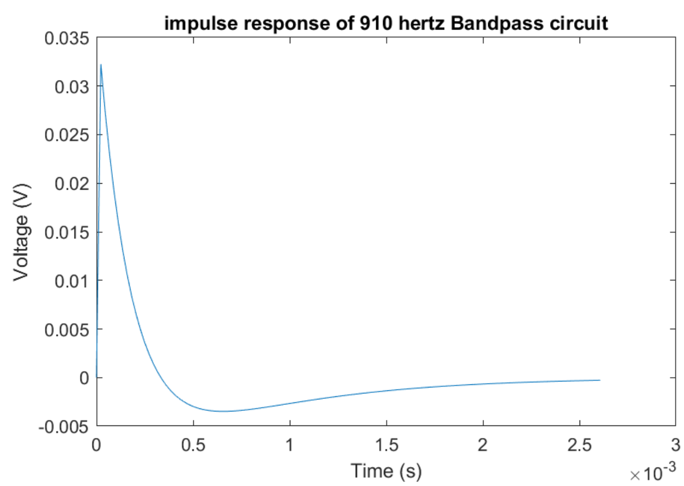


1. Low pass filter with a f_c of 230 Hz

Another step as a precaution was seeing what the frequency response was and if it was logically sound. The figure below shows the frequency response for a high pass at different frequencies, the high pass frequency behavior was ideal for a high pass.

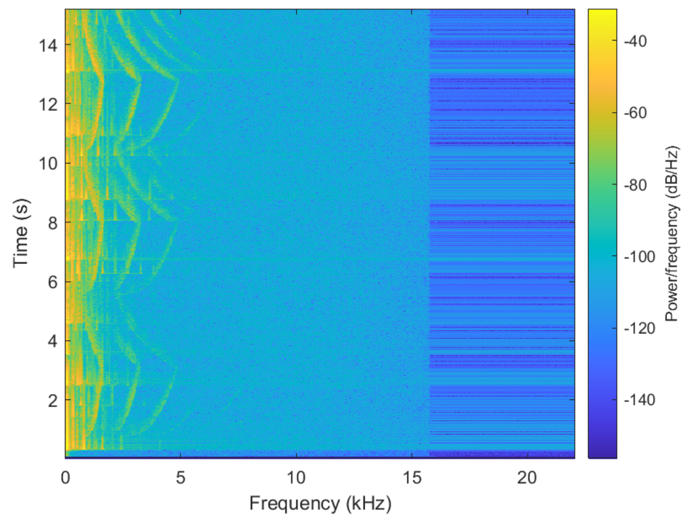


2. High pass filter frequency repsonse



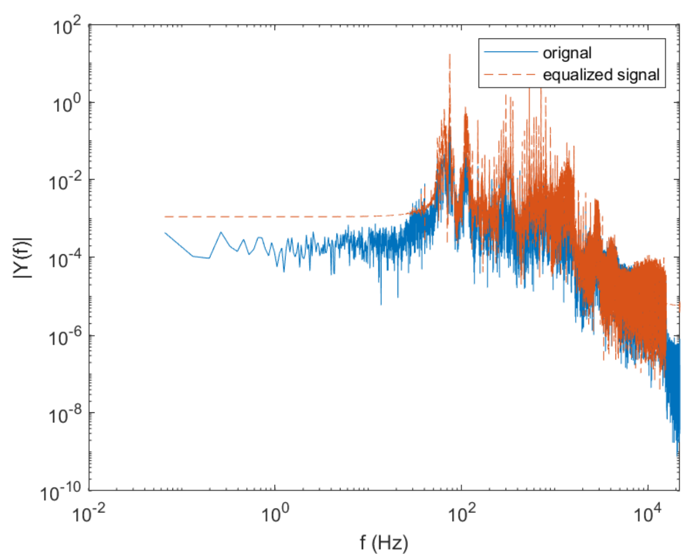
3. Impulse Response of 910 hertz Bandpass

   One of the most important part of this case study was constructing a bandpass by cascading a highpass to a low pass. One was to see if the bandpass is working correctly is by checking its impuls response. Above in figure 3 , the impulse response of the Banpasss can be seen and is the exact behavior of a bandpass.



4. Spectograph of filtered signal

   The method used to remove noise from the audio file was to see where the noise was located. As shown above the noise was at 5 frequency and lower. Therefore the equalizer was manipulated to reduce this noise and other frequencies were elevated. In addition, other bands had to be fixed in order to retrieve the original signal. The results for removing this noise and obtaining the original signal can be seen below



The results show that frequencies up to 5 hertz were filtered out. There was a great attempt to retrieve the original signal, but it slightly varied.

Overall, we were able to filter out noise by using the method above. For fixing presents it is important to understand jazz or the different types of music and what frequency it is played at in order to fix your presets so that it sounds better.

Conclusion

Throughout this case study, the power of signal processing and filters were successfully used to transform an input signal into the desired signal that the user wants. The use of RC circuits was shown to be effective in creating low pass, high pass, and band bass filters that successfully attenuated parts of the signal that were not desired due to noise or any other external reasoning. Furthermore, when putting multiple of these filters together across varying cut off frequencies, we were able to create an effective equalizer. By using the Equalizer, a listener can lower or raise the gain on the different frequency bands of the equalizer, thus making it easy and simple for even the average user to interact and alter the signal to their desired output. By taking the complex mathematics underneath the Fourier transform and packaging into a simple yet powerful interface such as an equalizer, we demonstrated how signals engineers deliver such value to the masses.

To take this design to the next level, automating the process by which the cutoff frequency is found would make the system more efficient and faster. To accomplish this, more research in machine learning and a larger data set of signals should be used to create an algorithm that can distinguish between the desired signal and the noise/unwanted frequencies. This is what happens in most modern noise cancelling headphones and is part of a multibillion-dollar industry.

References

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5. Equalized Signal Vs. Original