Overview of Filters Used in Parametric Equalizer

By Arnav Gupta, ECE '22

Introduction

In audio signal production and processing, there is a process known as equalization. Equalization is the audio process of adjusting volume/intensity of different frequency ranges of an audio signal in order to achieve a desired sound. An equalizer (EQ) is the type of device/filter that is used to complete this process. This process is what allows members of professional audio circles to transform their work into art and craft the exact sound they want. Some examples of equalization are changing the tones of instruments in a track or even increasing the bass on your car stereo. There are many kinds of EQ's that are used in audio processing but one of the most commonly used is a parametric EQ.

Background Information

Parametric EQ's are commonly used because they are the most flexible and controllable type of EQ. This is because parametric EQ's allow you to control the 3 most important parameters of equalization. These types of EQ's are also popular because they can be well represented and controlled by computer programs known as digital audio plugins. This is important because it allows both advanced, professional users and novice level users to utilize the device. Parametric EQ's typically offer control over 3-7 different frequency bands. These "bands" refer to a range of frequencies affected by a certain filter of the EQ. This is useful because every instrument plays at its own frequency. Having a range of bands is good because it allows you to control every part of your audio signal. Each band is controlled by a filter, but instead of coffee grinds being taken out by a coffee filter these filters allow you to affect certain frequencies while leaving others intact. For example, you could have an audio signal that has really high pitched (high frequency) instruments and low pitched instruments that are playing at the same time. Let's say you did not want the high pitched instrument to be as loud, the filter would let you reduce the volume of those instruments while leaving the low pitched instruments the same. A parametric EQ usually has 3-7 bands/filters that allow for a large amount of control over your audio signal and is why they are so commonly used by audio producers.

Controlling Parameters

Each filter in the parametric EQ has 3 controlling parameters that you can freely adjust. This is what allows you to control each filter to such a high degree. The 3 parameters are the center frequency, gain, and quality factor (Q). The simplest of these 3 parameters is the center frequency. In figure 1, you can see 6 lines/shapes. Each one of these lines/shapes represents a filter. The center frequency allows the middle of those lines/shapes to shift anywhere in a range of frequencies. The second parameter that can be controlled is the gain (represented by magnitude in figure 1). By increasing the gain you are making the peak of the lines/shapes higher and when decreasing the gain the peaks get lower. Setting the gain to 0 means that there will be no effect on the audio signal; you will get out what you put in. However, if the gain is greater than 0 the filter will increase the volume of the signal and if it is less than 0 it will decrease the volume of the signal. The effect of gain (either decrease or increase) will be most strongly seen at the center frequency and adjacent frequencies will be affected to a smaller degree. However, the effect of the filter on adjacent frequencies to f_c is not only affected by the gain but also the quality (Q) factor. This Q factor allows you to control the width of the filter. If you took the filters in figure 1 and increased their Q factor, the shape would change and form a steeper slope to the peak from both sides and affect a smaller range of frequency values. The opposite happens when decreasing the Q factor.

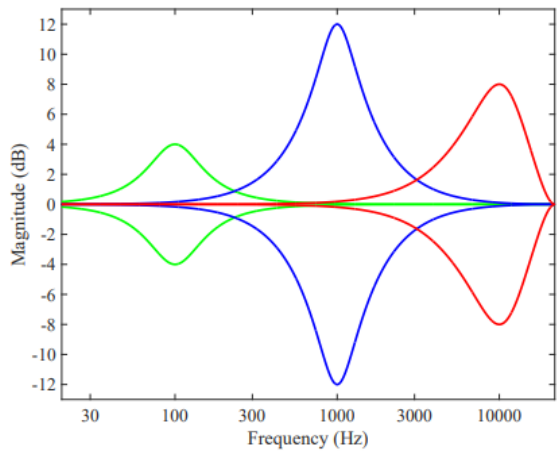

Figure 1: Magnitude responses of filter at f_c = 100 Hz (green), 1000 Hz (blue), 10000 Hz (red) with Q = 1

These controlling parameters allow you to adjust your audio signal to your liking.

Filters

There are 3 filters that are commonly used in parametric EQ's. These are the first order shelving filters, second order shelving filters, and second order peaking/notching filters. The shelving filters are not controlled by the same parameters as stated previously. These filters do not have any Q factor control and instead of having a center frequency they have a crossover frequency. The reason for this has to do with the shapes of these kinds of filters. An example of first order shelving filters can be seen in figure 2.

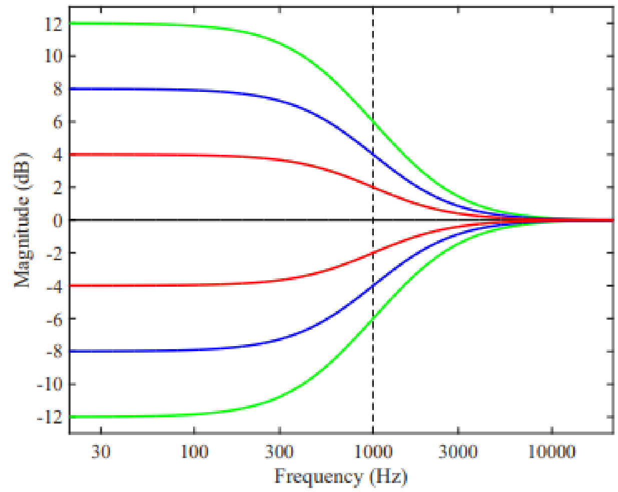

Figure 2: Magnitude responses of first order low pass shelving filter with ±12dB (green), ±8dB (blue), and ±4dB (red) with f_c set to 1kHz

As shown in figure 2, the first order shelving filter has no width to control. It simply increases/ decreases the gain of everything before or after the dotted line which represents the crossover frequency. This filter would be useful if you wanted to do something like increase the volume of all your low pitched instruments but leave the higher pitched ones the same. The second order shelving filter is almost the exact same as the first order shelving filter except that it has a steeper slope around its crossover frequency. This can be seen in figure 3.

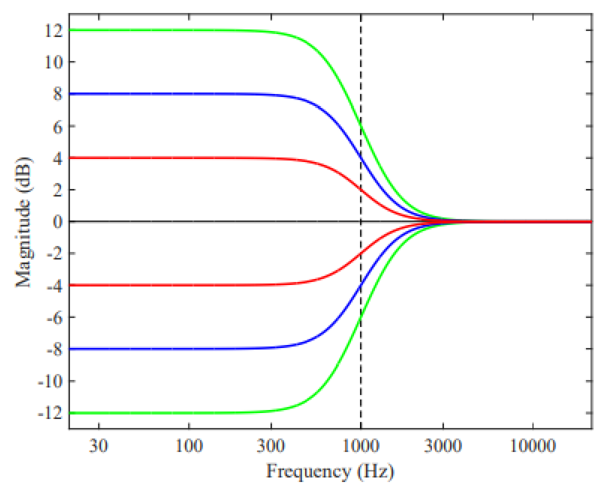

Figure 3: Magnitude responses of second order low pass shelving filter with ±12dB (green), ±8dB (blue), and ±4dB (red) with f_c set to 1kHz

While these filters do reach the desired gain in a shorter amount of frequencies, they are often more complicated to implement from a hardware standpoint. The final filter is the second order peaking/notching filter. Examples of these kinds of filters can be seen in figure 1. The goal of these filters is to be able to increase/decrease the gain of very specific frequencies. This is useful if you have unwanted noise in a middle frequency/pitch but don't want to affect the instruments at higher/lower pitches. You could also use it to boost an instrument at a middle pitch without affecting adjacent frequency pitched instruments. The combination of these filters allows an audio producer to tune their sound to exactly what they desire.

Conclusion

Parametric EQ's are extremely useful tools for professional audio producers. The ability to apply multiple filters at different frequencies and adjust the center frequency, Q, and gain of these filters allows the user to have a lot of control over their audio. Parametric EQ is useful because it employs various first order shelving filters, second order shelving filters, and second order peaking/notching filters in order to achieve this effect.

References

[1]. PreSonus. "What Is a Parametric Eq?" *PreSonus*, https://www.presonus.com/learn/technical-articles/What-Is-a-Parametric-Eq.

[2]. Arthur. "The Complete Guide to Parametric Equalization/EQ." *My New Microphone*, 8 Feb. 2022, https://mynewmicrophone.com/the-complete-guide-to-parametric-equalization-eq/.

[3]. Välimäki V, Reiss JD. All About Audio Equalization: Solutions and Frontiers. *Applied Sciences*. 2016; 6(5):129. https://doi.org/10.3390/app6050129

[4]. Orfanidis, S.J. IIR digital filter design (Chapter 11). In Introduction to Signal Processing; Prentice Hall: Englewood Cliffs, NJ, USA, 1996; pp. 573–643.

[5]. . Zölzer, U. Equalizers (Chapter 5). In Digital Audio Signal Processing, 2nd ed.; Wiley: Chichester, UK, 2008; pp. 115–19