Review of Electrical Filters

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Abstract

In electrical circuit theory, a filter is an electrical network that changes the amplitude and/or phase characteristics of a signal with respect to frequency. Thus a filter is designed to modify, reshape, or reject all unwanted frequencies of an electrical signal and accept or pass only those frequencies that are needed by the circuit designer. A common need for filters is in high performance stereo systems where certain ranges of audio frequencies need to be amplified or suppressed for best sound quality and power efficiency. Another well-known application of filters is in the conditioning of voltage waveforms in electrical power circuits. This manuscript provides a comprehensive review of the operation, characteristics, categories, and applications of filters.

Keywords: analog, digital, power electronics, digital signal processing, passive filter, active filter.

1. Introduction

A passive filter utilizes RL and RC circuits as its main components while active filter utilizes operational amplifiers as its main component. An example of a basic type of passive filter is shown in Fig. 1:
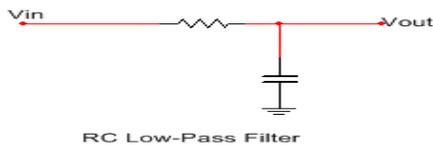

RC Low-Pass Filter

Fig. 1 RC low-pass filter circuit

There are some differences between passive and active filters. A simple example of an active filter is shown in Fig. 2:
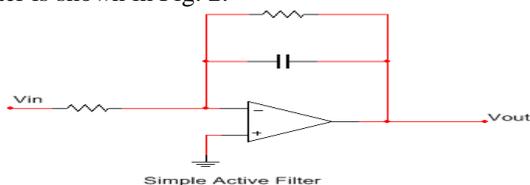

Simple Active Filter

Fig. 2 Simple active filter circuit

As it can be seen from the above example, active filters do not utilize inductors. Inductors are expensive and hard to fit into more discrete circuits. Instead, active filters utilize operational amplifier circuits which are cheaper to manufacture and are able to easily fit into discrete circuits. Another advantage of active filters utilizing operational amplifiers is that the circuits are able to produce desired cutoff frequencies with more precision than passive filters [1]. The aforementioned advantage is due to the ability to make higher order circuits using operational amplifiers. Along with having more precision in frequency selection, active filters allow for a very large range of ways to select the desired frequencies to be passed or blocked.

Filters are used in numerous applications. Among these applications are harmonic current compensation, reactive current compensation, voltage flicker compensation, voltage sag compensation, and negative phase sequence current compensation [3]. The aforementioned applications are primarily for power correction but filters have other applications as well. Other applications include digital signal processing and digital picture processing [10,11]. Digital signal and digital picture processing include digital and analog filtering. All these applications are found in industry such as: the power distribution industry, marine operations and the medical field to name a few [3,4,5,6].

Now that a brief overview of active and passive filters has been provided, a more detailed look into the aforementioned filters will be presented. The further explanation of active and passive filters will include more detail on active and passive filters. Following a description of operation of filters, their applications will be discussed.

2. Filters

This section presents a more in depth description of both active and passive filters. Content provided here includes a coverage of the operation of active and passive filters.

2.1 Passive Filters

Passive filters utilize both resistive-capacitive (RC) and resistive-inductive (RL) circuits. By utilizing the RC and RL circuits in different ways, one is able to construct four basic kinds of passive filters. The four different types of passive filters are low-pass, high-pass, band-pass, and band-stop [2].

A low-pass filter allows signals of a lower frequency to pass from the input to the output. The frequency of the filter can be determined by using the following formulas:

$$f_c = \frac{1}{2\pi RC} \text{ for RC } \quad (1)$$

$$f_c = \frac{1}{2\pi L/R} \text{ for RL } (2)$$

where R is the resistance in the circuit, C is the capacitance in the circuit, and L is the inductance in the circuit [2]. The low-pass filter can be made by using RC and RL circuits. Examples of these circuits are shown here:
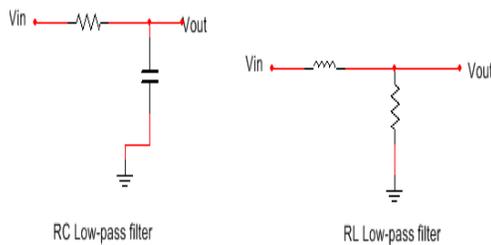

Fig. 3 RC low-pass and RL low-pass filter circuits

In the RC circuit, the output voltage is obtained from across the capacitor. While in the RL circuit, the output voltage is taken across the resistor. When the output voltage is 70.7% of the maximum voltage or higher, then the range of frequencies that make it happen will be allowed to pass [2]. The output voltage can be calculated using the following equation:

$$V_{out} = .707 V_{in} \quad (3)$$

In this case, the range of frequencies below the critical frequency, which is the frequency that will allow the output voltage to be 70.7% of the maximum voltage, will be allowed to pass, hence the name low-pass filter. In order to find the phase angle of the circuit, use Eq. (4) for the RC and Eq. (5) for the RL low-pass filters.

$$\theta = -\tan^{-1} \frac{R}{X_c} \quad (4)$$

$$\theta = -\tan^{-1} \frac{X_L}{R} \quad (5)$$

Where R is the resistance of the resistor and $X_C = \frac{1}{2\pi fL}$ and $X_L = 2\pi fL$. L is the inductance of the inductor and f is the frequency of the circuit.

The high-pass filter allows signals of a higher frequency to pass from the input to the output. Like low-pass filters, the high-pass filter can also be made by utilizing RC and RL circuits.
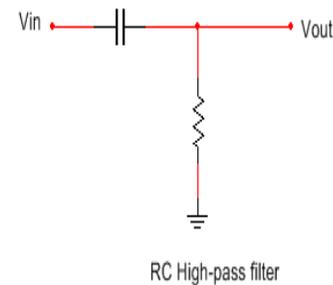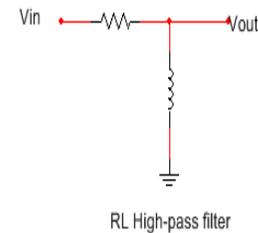

Fig. 4 RL high-pass and RC high-pass filters

In the RC circuit the output voltage is obtained across the resistor, while in the RL circuit, the voltage is taken across the inductor. Like the low-pass filter, when the output voltage is 70.7% of the maximum voltage or higher, the range of frequencies that permit the output voltage to achieve that level will be allowed to pass [2]. To calculate the output voltage of a high pass passive filter, one can utilize Eq. (3). In this case, all of the frequencies above the critical frequency will allow the output voltage to be greater than or equal to 70.7% of the maximum voltage. In order to compute the critical frequency, Eq. (1) can be used if the circuit is RC and Eq. (2) may be used if the circuit is RL. To calculate the phase shift, Eq. (6) is used for RC circuits and Eq. (7) is used for RL circuits:

$$\theta = \tan^{-1} \frac{X_C}{R} \quad (6)$$

$$\theta = \tan^{-1} \frac{R}{X_L} \quad (7)$$

Another type of filter is known as the band-pass filter. The band-pass filter utilizes a combination of high-pass and low-pass filters in order to get the desired output [2]. A band-pass filter does not just filter out high or low frequencies, it allows a band of frequencies to pass. The band of frequencies that are allowed to pass are the ones that are equal to and between the critical frequencies of the low and high-pass filters which constitute the bandpass filter. Therefore, Eq. (1) can be used to calculate the output voltage. To calculate the band width of the band-pass filter, which is the size of the band of frequencies that the filter allows to pass, the following equation can be used:

$$BW = f_{c2} - f_{c1} \quad (8)$$

where f_{c2} is the upper cutoff frequency and f_{c1} is the lower cutoff frequency [2].

Band-pass filters utilize both parallel and series circuits. Some examples of these circuits are shown here:
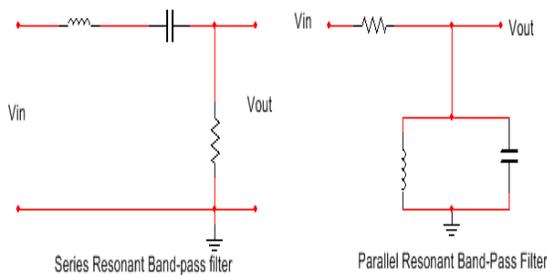

Fig. 5 Series resonant and parallel resonant band-pass filter circuits

When utilizing the parallel resonant band-pass filter, the inductor has the most impedance when the circuit is at its center frequency. When the inductance is at the maximum point, the output voltage will be at a maximum. Therefore, when the frequency is below or above the center frequency, the impedance decreases which allows for more voltage to be across the resistor in turn, causing the output voltage to decrease [2]. The range of frequencies being allowed to pass are the ones that allow the input voltage to be at or higher than 70.7% of the input voltage. When utilizing the series resonant band-pass filter, the circuit has the minimum impedance and the maximum current at the center frequency. As the input frequency begins to deviate from the center frequency, the impedance increases which causes the output voltage to drop [2]. However, in order to determine the band width of the resonant band-pass filter, a separate equation is given as:

$$BW = \frac{f_0}{Q} \quad (9)$$

where f_0 is the center frequency (resonant frequency) and $Q = \frac{X_L}{R}$. X_L is the reactance of the inductor, which can be calculated as $X_L = 2\pi f L$ where f is the frequency of the circuit and L is the inductance of the inductor within the circuit [2]. The filter only allows the band of frequencies to pass that make the output voltage 70.7% of the input voltage or higher. Therefore, the circuit allows for a band of frequencies to pass, making the circuit a band-pass filter.

The final type of passive filter is termed band-stop filter. This kind of filter is the exact opposite of the band-pass filter. Instead of having a band of frequencies to pass, the filter has a band of frequencies to stop. The band-stop filter utilizes both parallel and series circuits [2]. An example of these filters can be seen below.
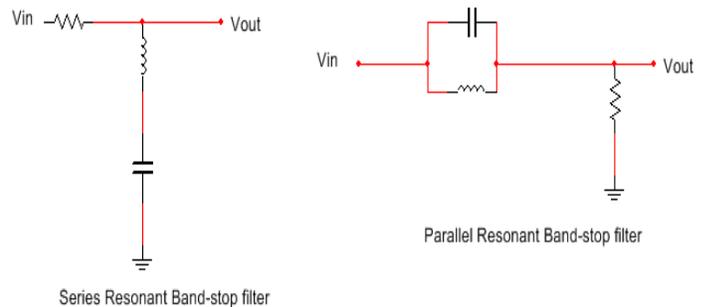
Fig. 6 Series resonant and parallel resonant band-stop filter circuits

A final characteristic to calculate for all filters is the gain of the filter. The following equation can be used for all passive filters and the equation is written as:

$$\text{Gain in decibels} = 20 \log(\frac{V_{out}}{V_{in}}) \quad (10)$$

2.2 Active Filters

An active filter is a filter that has four main types. The four types are the low-pass, high-pass, band-pass, and band-stop filters. These filters have the same functions as listed above for passive filters, except instead of utilizing inductors within the circuits, active filters use operational amplifiers. An example of an operational amplifier circuit is shown below.

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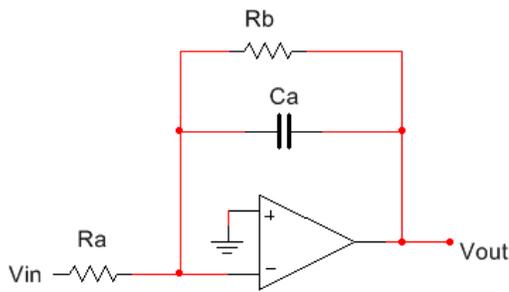
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Fig. 7 Simple active filter circuit 1   [1]

In order to determine the output voltage of the simple active filter circuit, the following equation should be used:

$$V_{out} = V_{in} \frac{-R_2}{R_1} \frac{1}{1+j\omega R_2 C} \quad (11)$$

where j is equal to $\sqrt{-1}$, ω is the frequency, R_2 and R_1 are the resistive values of the designated resistor and C is the capacitance of the capacitor [1].   To calculate the phase shift of a low-pass, high-pass, band-pass, or a band-stop active filter, the following equation is utilized:

$$\emptyset (\omega) = \tan^{-1} \frac{\omega}{\omega_0} \quad (12)$$

where ω is equal to frequency and ω_0 is equal to center frequency.   In order to find the gain of an active filter, Eq. (10) from the previous section can be used for any active filter. Although active filters seem similar to passive filters, in terms of their functions, there are structural differences.

2.3 Comparison of Active and Passive Filters

Passive filters use no amplifying elements, but they do use inductors. Active filters do not utilize inductors, but they do utilize amplifying elements. The difference of the components for active and passive filters is the main physical difference. However, there are differences in the advantages and disadvantages offered by each filter type. Advantages of passive filters are that they are the simplest filters in regard to the amount of components needed.   They are not limited to bandwidth limitations and so they work well in high frequency applications as well as high voltage and current situations.   Another advantage is that the passive filters generate very little noise in the output signal [15]. Some disadvantages of the passive filter are that it cannot provide a signal gain, the input and output impedances can be at levels higher or lower than desirable for certain applications, higher order circuits can be hard to design and passive filters

utilize inductors which are expensive and bulky [15]. Advantages for the active filters are that they may have high input impedances and low output impedances,   which   are   desirable   in   some applications. Active filters have almost any arbitrary gain, they are easier than passive filters to design, they do not utilize inductors, they are not bulky, they can be fit into discrete circuits and they are able to achieve higher accuracy in obtaining the wanted frequencies for the filter to pass [14]. Although there are many advantages of active filters, there are some disadvantages as well. Some of these disadvantages are that they are limited by bandwidth limitations so they are unable to operate at higher frequencies. Another disadvantage is that the circuit will generate noise in the output which is undesirable in the applications of filters. [15]

3. Order of Filters

One way to classify filters is according to their order.   The order of the filter refers to how many components are utilized within the filter. Below are some examples of classification of filters according to their order.

3.1 First Order Filters

A first order filter is one in which the circuit utilizes only one reactive element. This type of filter can be useful if the precision of the cutoff frequency is not imperative.   An example of a first order low pass filter based on an operational amplifier is shown below.
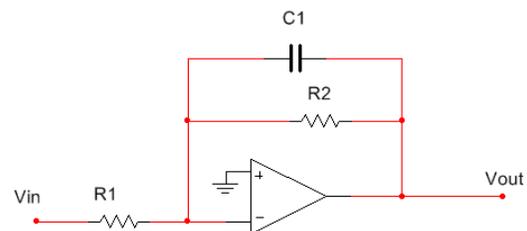

Fig. 8 First order low pass filter circuit with operational amplifier

For each of these filters, there are a number of calculations that are imperative to working with the circuit such as output voltage, the phase angle of the output voltage, and the gain of the filter.   For the low pass filter, the equation for the frequency response function is as follows:

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$$G(j\omega) = \frac{K}{(1 + \tau j\omega)} \tag{13}$$

where K is the gain, ω is the cutoff frequency, τ is the time constant of the system and j is $\sqrt{-1}$. The phase of the frequency response can be written as:

$$\arg(G(j\omega)) = -\tan^{-1}(\tau\omega) \tag{14}$$

where the variables are the same as in the previous equation. The voltage coming out of the filter can be written as:

$$V_{out} = \frac{I}{Cj\omega} \tag{15}$$

where C is the capacitance of the circuit and all other variables are as stated in the previous equations.

In regards to high pass first order filters, the frequency response equation is as follows:

$$G(j\omega) = \frac{K\tau j\omega}{1 + \tau j\omega} \tag{16}$$

where all variables are the same as previously stated. The phase of the output signal can be represented as:

$$\arg(G(j\omega)) = \frac{\pi}{2} - \tan^{-1}(\tau\omega) \tag{17}$$

The ratio of output voltage to the input voltage of the circuit can be written as:

$$G(j\omega) = \frac{V_{out}}{V_{in}} = \frac{RCj\omega}{1 + RCj\omega} \tag{18}$$

where R is the resistance of the circuit and C is the capacitance of the circuit. All the other variables are the same as in the previous mentioned equations. The band-pass filter equation for the frequency response function is represented as:

$$G(j\omega) = K \frac{1}{1 + \tau_1 j\omega} \frac{\tau_2 j\omega}{1 + \tau_2 j\omega} \tag{19}$$

where all variables are the same as in previous equations and the phase angle of the output can be shown as:

$$= \frac{\pi}{2} - \tan^{-1} \tau_2\omega - \tan^{-1} \tau_1\omega \tag{20}$$

where all variables are the same as previously stated. The ratio of output voltage to the input voltage of the filter is:

$$G(j\omega) = \frac{V_{out}}{V_{in}} = \frac{R_2 C_2 j\omega}{1 + R_2 C_2 j\omega \ \ 1 + R_1 C_1 j\omega + \frac{R_2}{R_1} R_1 C_1 j\omega} \tag{21}$$

The gain of a first order filter can be found as:

$$20\log \frac{Vout}{Vin} \tag{22}$$

3.2 Second Order Filters

A second order filter is the one that utilizes two reactive components to pass certain frequencies. The second order filters have more precision in the context of the desired cutoff frequencies. An example of a low pass second order RLC circuit is presented below:
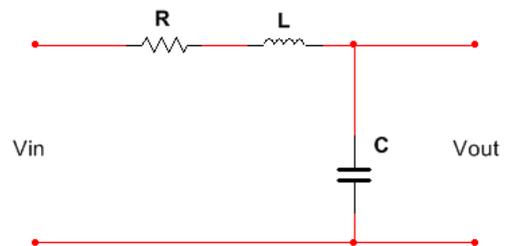

Fig. 9 RLC circuit

The equations for a second order low pass filter are shown below. The standard form of the second order low pass filter is represented as:

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$$T_{LP}(s) = \frac{T_{LP}(0)\omega_0^2}{s^2 + \left(\frac{\omega_0}{Q}\right)s + \omega_0^2}$$

(23)

where ω_0 is the pole frequency, Q is the pole or pole quality factor and s is the complex frequency variable. The ratio of output voltage to the input voltage of the filter is expressed as:

$$\frac{V_{out}(s)}{V_{in}(s)} = \frac{\frac{K}{(RC)^2}}{s^2 + \left(\frac{3-K}{RC}\right)s + \frac{1}{(RC)^2}}$$

(24)

where K is the gain, R is the resistance in the circuit and C is the capacitance in the circuit. The phase angle of this type of filter is:

$$\phi = \tan^{-1}\left[\frac{\sqrt{1-\zeta^2}}{\zeta}\right]$$

(25)

where ς is the damping factor which is given as $\frac{1}{2Q}$. The high-pass filter equation for the transfer function is:

$$T_{HPn}(s_n) = \frac{T_{HP}\left(\frac{s}{\omega_0}\right)}{|T_{HP}(\infty)|} = \frac{s_n^2}{s_n^2 + \frac{s_n}{Q} + 1}$$

(26)

where S_n is the complex frequency variable given as $\frac{s}{\omega_0}$. Vout and Vin can be solved for using the given equation by setting Vout over Vin equal to the transfer function and then solving for the desired variable. The band-pass filter's second order transfer function can be written as:

$$T_{BP}(s) = \frac{T_{BP}(\omega_o)\left(\frac{\omega_o}{Q}\right)s}{s^2 + \left(\frac{\omega_o}{Q}\right)s + \omega_o^2}$$

(27)

where ω_0 is the pole frequency, Q is the pole quality and s is the complex frequency variable. For any second order filter, if the gain of the filter is needed, one can utilize the following equation:

$$20log\frac{Vout}{Vin}$$

(28)

3.3 Applications of Different Order Filters

When using different ordered filters, there are a number of different aspects to consider when choosing what order filter to utilize in a specified application. Some aspects to consider when choosing a specific order filter are budget, accuracy of cutoff frequency, size restrictions and time constraints. As the order of the filter increases, the more expensive the filter becomes. The rise in cost is due to the increased amount of components needed to design the circuit. Therefore, if a project needs an electrical filter and the budget is tight a higher order filter may not be the best choice. Not only does cost increase as filter order increases, but the accuracy of the filter's cutoff frequency increases up to a certain point at which the accuracy then decreases. This is because as the filter order increases, the falling slope at the cutoff frequency increases by 20dB/decade. Therefore, as the slope increases, the range of frequencies that pass after the cutoff frequency diminish until the slope reaches a certain point where the slope would then cut into the desired pass frequencies. Size restrictions become a prominent concern as higher order filters are used. Since more components are being utilized, more space is needed by the filter. If a filter is being designed for a tight space, then a lower order active filter would be the best choice based only on size. A final basic consideration when deciding what order filter to use is based on time constraints. If the time in which the filter is needed to be designed is little, then a lower order filter would be useful, however if time is not an issue then a higher order filter could be used. The higher the filter order, the longer it is for the engineer to successfully design the filter to the required specifications.

4. Digital Filters

A digital filter is essentially a discrete-time, discrete amplitude convolver [25]. Digital filters can be used in multiple applications and they can be designed in many ways to fit the desired application. Among the many ways to design a digital filter, there are also multiple types of digital filters that can be implemented into a design. In designing the filter, there are also different algorithms used to make the filter function in the desired manner.

4.1 Digital Filter Fundamentals

When analyzing a digital filter, one looks at it in frequency domain. The digital filter is

represented in frequency domain because frequency domain allows someone to look at how the time domain impulse affects the shape of the sampled signal. When designing an analog filter, the filters are designed with a frequency domain response that is as close to the ideal response as the filter will allow [25]. Designing the frequency domain response accurately according to the desired parameters is crucial because this response gets transformed into a time domain impulse. The time domain impulse gets transformed into the coefficients of the filter. The coefficients of the filters are what allow the filter to act in the desired manner. The coefficients of the filter are not the only thing that has an effect on the function of a filter, it is also the device that sends the signal to the filter. Depending on the the order and type of polynomial used to approximate the ideal response and the number of bits used to calculate the math along with the filter, the filter's pass band ripple, shape factor, stop band attenuation and phase characteristics are affected. [25]

4.2 Design of Digital Filters

In order to better understand how digital filters are designed will be helpful in understanding the characteristics of the different types of digital filters. In designing an IIR filter, one would need to choose the model to follow. It may be a Butterworth, Chebyshev or Elliptic filter. The type of filter is chosen based on the desired properties for the filter. Once the prototype of the filter is chosen, the method of how to transform the filter into a digital filter is selected. Generally for IIR circuits, one can transform the circuit by impulse variance or by bilinear transformation. Once the filter is in the digital form, the frequency can also be manipulated in order to make the filter band pass or high pass as desired. [29]

4.3 Types of Digital Filters

When studying digital filters, there are two main types, finite impulse response (FIR) and infinite impulse response (IIR). The key difference between the two filters is that the FIR filters have a no non-zero feedback coefficient while the IIR has one or more non-zero feedback coefficients. Not having a non-zero feedback coefficient means that when the filter is energized by a signal, there is only an output for a certain number of computational cycles. Having a non-zero feedback coefficient means that when the circuit is energized by a signal there will always be a response. [30] In addition to whether or not the filter has a non-zero feedback coefficient,

there are other characteristics that set the FIR and the IIR apart from one another. The IIR has nonlinear phase characteristics and is able to make sharper transitions. The IIR filter is also difficult to use when the application requires the filter to be adaptable. The difficulty of applying IIR filters to adaptable applications is due to the IIR filter's feedback loop. [30]

There are many other types of digital filters that can be made from the aforementioned filters. A few simple types of digital filters that can be implemented into the FIR and IIR filters are the low pass, high pass, band pass and band stop filters. In addition to the previously stated types of filters, there are also Butterworth, Chebyshev, and elliptic filters. [30]

Butterworth filters have two changeable characteristics, the order of the filter and the cutoff frequency. In relation to the changeable characteristics, as the frequency increases, the magnitude response of the filter decreases. When designing the Butterworth filter, one needs to be careful because when the filter is implemented in the direct form, the filter may cause instability within the circuit. To resolve this issue, one can combine first order and second order Butterworth filters into one third order filter. [30]

Chebyshev filters have three changeable characteristics when designing the filter. The three characteristics are order, pass band cutoff frequency, and the pass band ripple where the maximum gain is 1 and the minimum is 1 minus the pass band ripple. Chebyshev filters have two types, one and two. Type 1 filters are equiripple in their pass band, but they decrease monotonically in the stop band. In type 2, the filter is monotonic in the pass band and equiripple in the stop band. In designing type 1 and type 2 Chebyshev filters, if there is a monotonic characteristic in the pass band or stop band then that suggests that the system consists of a low order system. Whether or not the system is low order varies depending on whether or not the equiripple is in the passband or stop band. [30]

Elliptic filters allow for the designer to change four characteristics consisting of pass band ripple, stop band ripple, order and pass band edge. When designing an elliptic filter, the order of the filter controls the transition bandwidth. An important characteristic to take into consideration when designing an elliptic filter is that it is equiripple in the pass band and stop band. [30]

In relation to every filter mentioned above, each of the filters has a non-constant group delay due to the non-linear phase. The greatest group delay occurs at the edge of the pass band or within the

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transition band. Overall, the Chebyshev type 2 filter has the least delay in the pass band and the largest region of the pass band in which the delay is constant. If the group delay does not need the delay to be constant, then the elliptic filter approximation yields the lowest order system function. [30]

4.4 Advantages/Disadvantages of Digital Filters

In relation to analog filters, digital filters have advantages and disadvantages. Some of the advantages that digital filters have that they offer more flexibility, have a higher performance in attenuation and selectivity, offer an improved time and environment stability and they are lower in cost to produce. A disadvantage of digital filters is that they are limited by how much processing power can be put into the filter.

4.5 Applications of Digital Filters

When considering the applications of digital filters there are multiple characteristics that the circuits can output using certain optimizations. Some of the characteristics include low speed and low power, moderate speed and moderate power, and high speed applications. If a low speed and low power application is needed, one would want to use a bit-serial implementation. When a moderate speed and moderate power application is to be developed, one would need to use a single parallel multiplier accumulator. For a high power application, a multiple multiplier-accumulator structure with specialized memory address schemes is needed.   When optimizing decimating or interpolating filters, one would want to use polyphase structures and specialized memory addressing systems. An example of digital filter applications consist of audio and high quality image processing which would include speakers and digital converter boxes for televisions. [25] Alongside image and audio processing, there are other digital filter applications such as, dynamic sensor compensation and parametric equalizer design [26, 27].

4.6 Algorithm Types

When working with digital filters, there are multiple types of algorithms that can be used to make the filter run in a desired fashion. Common types of algorithms include recursive algorithms and implicit recursive algorithms [28]. In recursive algorithms, there are multiple types of algorithms that include exponential, hyperbolic and elliptical. In addition to the recursive algorithms, there are a plethora of other

algorithms that can be utilized to help make a filter function in the correct manner. Some of these algorithms are the Remez exchange algorithm, Fletcher –Powell algorithm, efficient digital signal processing algorithm and adaptive algorithms. Adaptive algorithms have two popular kinds, the least mean square algorithm and the recursive least square algorithm [31].

5. Analog Filters

After a discussion of digital filters, the other type of filter is the analog filter. The analog filter acts as a kind of template for creating the digital filter. Analog filters are designed first with the specified characteristics and then converted to digital filters [32].   In addition to the use of passive components, an analog filter has an input signal that is analog as well as an output that is analog instead of having a digital output like a digital filter [33].

5.1 Analog Filter Fundamentals

As mentioned in the previous section about digital filters, digital filters are implemented in frequency domain. Frequency domain is the form in which a analog filter is designed. The only difference between a digital filter and an analog filter is that after the filter is designed in frequency domain, the digital filter gets converted into digital form while the analog filter is kept in frequency domain.

5.2 Design of Analog Filters

When designing analog filters, there are a few decisions to be made in relation to the kind of filter that one wants to use. The first decision is whether to have the signal rippled or flat within the pass and stop bands. Different kinds of filters make the signal flat or rippled in either the pass or stop band. For example, Butterworth filters are flat in both bands, Chebyshev type 1 filters are rippled in the passband and flat in the stop band. Chebyshev type 2 filters are flat in the passband and rippled in the stopband and elliptical filters are rippled in both bands. When deciding whether or not to have a specific band flat or rippled, we need to take into consideration the fact that more ripples band has, the narrower the transition band will be. Also when designing filters, sometimes one needs to transform the original filter into a different type of filter. An example of a transformation would be going from a low pass analog filter to a high pass or some other kind of filter.

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5.3 Types of Analog Filters

Within the realm of analog filters, there are three different types of analog filters that can be utilized. The three different kinds are network synthesis filters, impedance filters, and simple filters. In the previous section two kinds of filter was already explained and that was the network synthesis filters and the simple filters.

The network synthesis filters include the Butterworth filter, Chebyshev filter, Elliptic filter, the Bessel filter, Gaussian filter, Optimum Legendre filter, and the Linkwitz-Riley filter [35]. The only three types of filters in this group that are not discussed earlier are the Bessel filter, Optimum Legendre filter, and the Linkwitz-Riley filter. The Bessel filter is a filter that is optimized to have a maximally-flat time display which then leads to the filter having a linear phase response. The Bessel filter is the only network synthesis filter to have a linear phase response [34]. The Optimal Legendre filter is a great application in microwave filtering due to its unequal ripples in the pass band and the impedance matching bandwidth that is also in the pass band [36]. The Linkwitz-Riley filter is a filter that utilizes a low pass and a high pass filter in which are made by cascading either low pass or high pass Butterworth filters [37].

In the simple filters group it includes RL, LC, RC and RLC filters [35]. In the section two of the paper, RL and RC circuits were discussed but LC and RLC circuits were not. An LC circuit is the one that contains both an inductor and a capacitor and an RLC circuit contains a resistor, inductor and a capacitor.

The impedance filters group includes the constant k filter, m-derived filter, general image filters, Zobel network [35]. A constant k filter is a filter in which as the order of the filter increases, the response of the filter also increases. Another characteristic of constant k filters is that the same components have the same values for each piece of equipment [38].

The m-derived filter is one in which the attenuation in the stop band is very fast and when the filter is converted from prototype to actual product, the m-derived filter utilizes the same characteristics as the original prototyped circuit [39].

General image filters are filters that are used in order to filter image signals. They can be used often to increase the sharpness of the boundaries of an image so as to appear clearer in the output [40]. The final type of filter discussed in the impedance group is the Zobel network. The Zobel network is a

network that can be used in order to produce an output that would lead to it being resistive. This specific kind of network utilizes R, L and C in order to function. However, the circuit primarily uses resistors and capacitors [41].

5.4 Advantages/Disadvantages of Analog Filters

Analog filters have several characteristics which may be regarded as either advantages or disadvantages in the context of filter application [33]. Some of the key characteristics of analog filters are:

- They have less accuracy compared to digital filters.
- They have drift.
- Adaptive filtering is difficult.
- No digital to analog conversion is needed.
- No digital signal processing is needed.

5.5 Applications of Analog Filters

When using analog filters there are numerous applications for these devices. A few examples of these applications are to stabilize amplifiers, to block direct current offset in high gain amplifiers or single supply circuits, and to separate signals to name a few. In addition to the aforementioned applications, there are other products that these applications can be used in. Examples of real products that the previous applications can be used in are radio receivers, data conversion, and reconstruction of a signal at the output of a D/A. [43]

5.6 Comparison of Analog and Digital Filters

In addition to what was already said, there are other differences between analog and digital filters. One difference is that when plotting the frequency response plots of each type of filter, the digital filter is plotted in a linear frequency scale while analog is plotted logarithmically. Another difference is that the analog filter has a superior amplitude dynamic range than a digital filter would have. Digital filters are also slower than analog filters. A digital filter's step response is symmetrical while the analog filter's response graph is not symmetrical. In addition to the differences between the step response of an analog filter and a digital filter, there is also a difference in how the filter rolls off and how the filter attenuates the in the stop band. Another way in which analog and digital filters are compared is in terms of filter behavior in the pass band. Typically in an analog filter, the pass band

contains about 6% ripple while the digital filter is flat in the pass band. [51]

6. Power Electronics Application

6.1 Why are Filters Used in Power Electronics?

Knowing that power electronics efficiently converts electrical energy into the desired ranges for a specific machine or piece of electronics, filters are extensively used in power electronics devices. Filters are important because sometimes when the signal is put into the system, harmonics are created which are most of the time undesirable in electronics [44]. By placing a filter at the input of the power electronics, the noise and harmonics can be filtered out so that the system can convert the signal into whatever characteristics of the signal are required. Not only can filters be placed on the input of the power electronics, but they can also be on the output as well. In some cases, the signal going in needs to be filtered as well as the signal going out. Sometimes the filtered signal has an undesired spike. In order to counteract the spike, the filter can be damped [44]. By damping the filter, the spike will be diminished and the filter will meet the requirements of power electronics system.

6.2 How are filters for Power Electronics Designed?

There are several steps involved in the design of a filter for power electronics applications. Although there are different ways in which to obtain the design of a filter, there is one characteristic that one practically always wants in a filter for power electronics. The characteristic is to have the change in signal be as fast as possible [45]. In order to achieve the quick signal change, one needs to pay close attention to the time-domain characteristics [45]. Time-domain dictates how quick the signal will be filtered. In addition to the main characteristics of all types of filters, the specific procedures to follow for the electromagnetic interference (EMI) filter will be discussed. To design the EMI filter, one needs to know three specific pieces of information. The first is to know that impedance has little effect if the filter's components are in proper order and size. It is also important to know how much EMI noise will be produced that needs to be filtered. Finally, it is important to know that when designing the filter, the main goal is to meet the low-frequency requirement of the system. [46] Once these pieces of information

are realized, then the design of the filter can be developed. The first step is to measure the base-line values of the system. Base-line values will allow the designer to know what values need to be filtered into specified values. After the base values are known, the attenuation requirements need to be known. Certain signals need to be attenuated more than the others, but the amount of attenuation varies based on what application the signal will be used for. The third step is to determine the corner frequencies of the filter. Once the corner frequencies are determined, the filter component values are to be determined. [46] After all the values are determined the filter can be constructed and the EMI filter is complete.

6.3 Advantages of Filters in Power Electronics

When utilizing filters within power electronics, there are multiple advantages to do so. One advantage is that the signal going in and out can be set to designated parameters. By having the signal set to certain parameters, power electronics can be tailored to fit practically any application. Another advantage is that filters can help to keep other machines protected. In some cases when harmonics are produced from the signal, they may have an effect on other machines [44]. Depending on the harmonic and the machine that the harmonics are affecting they can have a detrimental effect on the function of the machine. A final advantage is that the time-domain of the filter can be changed [45]. As a result, the signal can be filtered at a high rate of speed. When the signal can be filtered quickly, the machine may run more quickly without having to wait for the signal to be filtered.

6.4 Disadvantages of Filters in Power Electronics

With advantages also come disadvantages. Although filters are very useful in power electronics applications, one disadvantage is that although the filter attenuates the signal to the desired parameters, the transfer functions for the converter within the system are changed. As a result of the transfer functions being changed, they have to be changed back again by using new transfer function equations [44]. Another disadvantage is that when trying to design certain filters to be specific parameters, such as the electromagnetic filter, it is often very hard to come up with the proper design [46]. Even when one does come up with the proper design, it often takes a very long time in order to get the correct values for the components of the filter.

7. Digital Signal Processing Application

In the digitally advanced society of today, many applications deal with some form of analog signal. When the analog signals are put into a digital application, the analog signal needs to be converted into a digital signal in order for the digital application to read the signal. Whenever the analog signal is converted, digital signal processing comes into play. The following information will explain how and why digital filters are used in digital signal processing. The advantages and disadvantages of using digital filters in digital signal processing (DSP) are also listed.

7.1 Why Filters are Used in Digital Signal Processing

Filters need to be used in digital signal processing applications because when processing signals, there will always be some unnecessary frequencies that will affect the quality of the signal. Having a quality signal will allow for a better quality output. Quality is key for many applications, such as, for sound and audio applications. When referring to quality it is not just how clear the signal is, but it also refers to how close to the desired parameters is the signal for the application it is being used for.

7.2 How Filters are Used in Digital Signal Processing

As stated above, filters are used for digital signal processing applications by filtering out unwanted frequencies. Whenever a signal is introduced to the system, it first goes through an anti-aliasing filter. This filter takes the input frequency and then cuts off anything higher than one half of the sampling frequency [48]. If frequencies above one half of the sampling frequency passes, would cause manipulation of the signal. Once the signal contains the correct frequencies, then the input analog signal is converted into a digital signal using an analog to digital converter. After the signal is converted, it sometimes needs a conversion once again to correct specifications. Depending on the technology being used, the signal may need to be converted back to an analog signal. [48] The process of which the signal is processed from input to output can be seen visually below:

Fig. 10 Process of digital signal processing

Knowing how digital signal processing uses filters, another key aspect of filter application is choosing what kinds of filters to use within the system. When choosing the filter to use, there are several types one could choose from. However, in this application, digital filters are mainly used instead of analog filters. The two main types of digital filters for this application are the FIR and the IIR filters [48]. Both these filters were explained previously. When choosing whether to use the FIR or the IIR one must look at the characteristics of each filter and choose the one that will suit the application. Every application has important specifications to meet and by knowing the specific characteristics of the desired application, one can accurately choose and decide the correct filter to use within the system.

7.3 Advantages of Using Filters in Digital Signal Processing

When reviewing the advantages of using filters in digital signal processing, the focus will be on the advantages of using digital filters. An advantage of using digital filters in digital signal processing is that it is simple to design filter circuit. When using analog filters, one must put together passive elements and because of this, it would take more time to build the circuit. However, when using digital filters, the designer only needs to put one chip in place and then program the chip to the desired parameters.

Another advantage is that when using digital filters, it is much easier to modify and calibrate the filters. Instead of having to add or replace software in analog filters when using digital filters, one can simply reprogram the digital filter to whatever specifications one needs it to be.

A final advantage to using digital filters in digital signal processing is that it is a very affordable way to provide filtering in an application where many analog components would be needed. Instead of purchasing many different components, one just has to purchase the chip with the filters on it. After purchasing the chip, one can develop any configuration of filter one wishes to use. [49]

7.4 Disadvantages of Using Filters in Digital Signal Processing

Although there are many advantages of using digital filters in DSP, there are some disadvantages as well. One disadvantage is that the cost of the digital signal processing system can be quite high [50]. In some applications, what the system needs to accomplish is so minimal that it would be more cost effective to buy the few analog components instead of purchasing a more expensive digital component. Another disadvantage is that, as mentioned previously, the digital filter is limited to how much the filter is able to process. Often in this situation, the more processing power, the more space the processor takes up. Therefore, in applications where the filter must fit into a small space, the processor would be limited. [33]

7.5 Digital Signal Processing Software Packages

When utilizing digital signal processing, there are a number of software packages that can be used when designing components to be used for digital signal processing. Two of these software packages are MATLAB and eXpressDSP. When choosing what kind of software package to use, one needs to know what each package has to offer.

When utilizing digital signal processing with MATLAB, it allows one to acquire, measure, and analyze signals from many sources. It also allows one to design streaming algorithms for audio, smart sensor, instrumentation, and wearable devices as well as allowing one to prototype, test, and implement digital signal processing algorithms on PCs, embedded processors, SoCs, and FPGAs. Along with the aforementioned abilities, MATLAB also allows one to preprocess and filter signals prior to analysis, explore and extract features for data analytics and machine learning applications, analyze trends and discover patterns in signals, and visualize and measure time and frequency characteristics of signals. Also when designing, MATLAB allows a designer to design and simulate streaming algorithms for audio, video, instrumentation, smart sensors, wearable devices and other electronic systems. [53]

EXpressDSP offers standardization and software reuse that allows the development of a product to be accomplished more easily. It specifically offers standards enabled for modular, reusable multi-function applications, allows a developer to integrate their own software, offers a catalog of interoperable software, it focuses on adding value/differentiation to the software, and it is designed to cut development time by 50 percent. Not only does eXpressDSP allow development to be easier, it also offers tools to reduce system cost, enhance product robustness and innovation as well as accelerating the time-to-market process. The software allows a user to accomplish the aforementioned objectives by offering powerful, integrated development environments, scalable real-time kernel, eXpressDSP compliant algorithms, reusable modular software and support. The software is able to run on OMAP application-specific processors and DaVinci digital media processors among others. It offers advanced data visualization and real-time analysis and powerful code generation tools. [52]

8. Conclusion

In summary, a filter consists of an electric circuit that takes an input signal and then allows certain frequencies to pass. These frequencies can either be on the low end, high end, or a range of frequencies. In order to achieve the desired frequencies, there are four main kinds of filters: low-pass, high-pass, band-pass, and band-stop. Filters are not only categorized by what frequencies they let through, but also by what kind of components they have. Filters categorized in this way are either active or passive. Passive filters include elements such as inductors, capacitors and resistors while active filters contain elements such as operational amplifiers. In addition to the previous ways of categorizing a filter, a filter can also be categorized by order. Order simply refers to how many filters are cascaded together. A filter can also be categorized by whether it is a digital or analog filter. Analog filters rely on actual components such as resistors and inductors to filter the signal while digital filters use chips that depend on some time code to filter the signal. This manuscript presents detailed information regarding different categories and applications of filters.

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