Design of ultralow noise and THD low pass filter for audio analyzer

by

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16073

Dissertation submitted in partial fulfillment of

the requirements for the

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CERTIFICATION OF APPROVAL

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Approved by,

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Dr Wong Peng Wen

UNIVERSITI TEKNOLOGI PETRONAS

TRONOH, PERAK

January 2016

CERTIFICATION OF ORIGINALITY

This is to certify that I am responsible for the work submitted in this project, that the original work is my own except as specified in the references and acknowledgements, and the original work contained herein have not been undertaken or done by unspecified sources or persons.

Justin Lee Kung Wen

Abstract

An ultralow noise and THD low pass filter is required to design for Keysight Technologies audio analyzer model U8903B. The company provides the required specifications for the filter. There are two different filters required based on the installed option. Passive low pass Chebyshev ladder network is constructed due to its characteristics. It is constructed by cascading capacitor and inductor. Air core inductor is used to minimize THD.

Acknowledgement

Firstly, I would like to thank my supervisor, Dr Wong Peng Wen and co-supervisor, Dr Lee Yen Cheong for helping, guiding and sharing their knowledge and experiences with me. Next, I would like to thank my friends for helping me in this project especially Cheab Sovuthy and Elden Zee. Lastly, I would like to appreciate the examiners that comment on the project.

TABLE OF CONTENTS

LIST OF FIGURES

LIST OF TABLES

Chapter 1: Introduction

1.1 Background

In signal processing, a filter is defined as allowing some certain frequency spectrum range to pass through and blocking the rest of the other frequency range. There are two types of filter that is commonly known which is analog filter and digital filter. Analog filter uses analog component to perform the filter while digital filter uses digital electronic which only uses two level and discrete band of analog level. Figures below show the filter design for digital and analog.

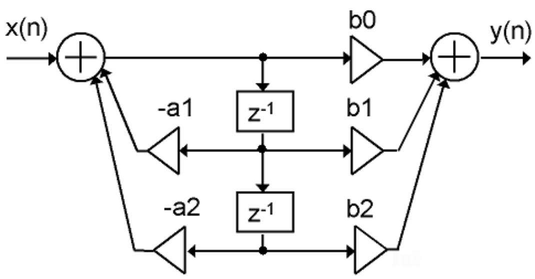


Figure 1: Digital filter

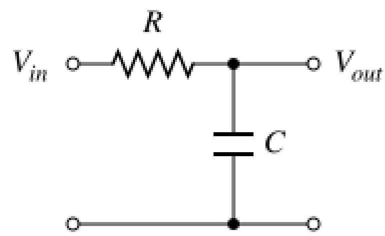


Figure 2: Analog filter

Every filter has its own specification, for example: transfer function and cutoff frequency. Transfer function is defined as the ratio of output to its input which is usually expressed in decibels to observe its relationship. Cutoff frequency is a parameter to determine whether frequency above or below this will be attenuate.

In analog filter, it is classified into low pass, high pass, bandpass and bandstop filter. Low pass filter allows low frequency to pass and high pass filter allows high frequency to pass. Bandpass filter allows frequency between two points to pass and blocks the remaining frequency while bandstop filter blocks the frequency between two points but allows the other frequency.

Besides, analog filter is categorized into passive and active filter. Passive filter uses component like resistor, inductor and capacitor which does not required external power source to operate. For active filter, it might uses a combination of passive and active component which required external power source like operational amplifier.

Power quality of a distributed system will affects the power consumption and regulation. According to Fuchs and Masoum (2015), power quality is defined as the measure, analysis and improvement of the bus voltage to maintain a sinusoidal waveform at rated voltage and frequency. Hence, power source itself is a nonlinear source which will draw distorted waveform that contains harmonic. Harmonic is a waveform produce by its fundamental frequency. For example, a waveform with fundamental frequency of 50Hz will have harmonic at 100Hz, 150Hz, 200Hz and so on as its second, third and fourth harmonic. The ratio of these total harmonic components to its fundamental frequency component is called total harmonic distortion (THD). Thus, THD can be defined as how much is the harmonic waveform is different form its fundamental waveform. The lower the THD, the lesser the interference towards other electronic equipment.

$$THD = \frac{\sqrt{V_2^2 + V_3^2 + V_4^2 + V_5^2 \ldots + V_n^2}}{V_1^2}$$

1.2 Problem Statement

There is a collaboration project with Keysight Technologies on audio analyzer. Hence, an ultralow noise and THD low pass filter is required for the sound analyzer. A specific inductance value is required for the low pass filter design but it cannot be obtained commercially.

1.3 Objective

To design a LC low pass filter and specific value of air core inductor for the sound analyzer. Air core inductor is constructed because it will have the least amount of distortion. The present of ferromagnetic material in the core will make the inductor to reach its saturation point easier. When the inductor reached its saturation point with a high current flowing through it, the inductance will not be constant and this will create distortion.

Chapter 2: Literature Review and Theory

2.1 Filter

The impedance of an inductor is given as $X_L = 2\pi fL$ while the impedance of a capacitor is given as $X_C = \frac{1}{2\pi fC}$. The figure below shows a LC low pass filter.

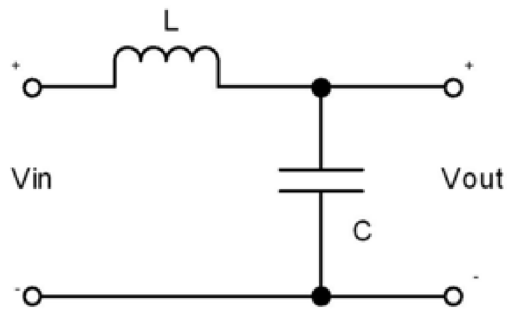


Figure 3: LC low pass filter

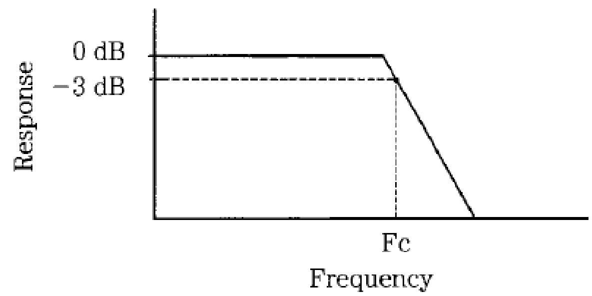


Figure 4: Low pass filter frequency response

When a low supply frequency is applied to the LC low pass filter, the impedance of the inductor is low while the impedance of the capacitor is high. Hence, the inductor can be considered as a short circuit while the capacitor is open circuit. As a result, the output will be the same as the input of the filter.

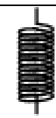
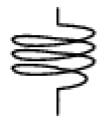
On the other hand, if a high supply frequency is applied to the filter, the impedance of the inductor will be very high and the input source of the filter will not reach its output.

The cutoff frequency can be calculated by $\frac{1}{2\pi\sqrt{LC}}$ or a -3dB at its filter frequency response. This is because -3dB is the result of half of its input present in the output. It can be calculated by $10\log(\frac{output}{input})$.

2.2 Inductor

2.2.1 Factors that affects inductance value

Table 1: Factors that affects inductance value

Factor	Less inductance	More inductance	Explanation
Number of turn			More turn will produce more magnetic field
Coil area			Bigger area of coil will have lesser opposition of the magnetic flux formation
Coil length			Longer coil length will have a higher resistance and result a lower inductance
Core material			Higher magnetic permeability material will produce more magnetic field

The formula for inductance is given by

$$L = \frac{N^2 \mu A}{l}$$

where N=number of turn
μ=permeability of core
A=cross sectional area of coil
l=length of coil

2.3 Two port network

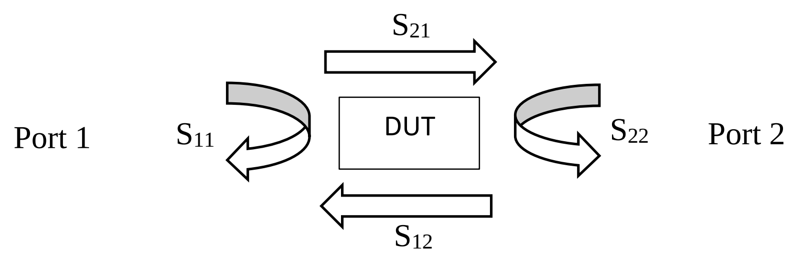


Figure 5: Two port network

Two port network is an electrical network with an input and output port. Scattering parameters is used to relate the transmission and reflection waves at each port. S_{11} and S_{22} are the reflection coefficient due to its incident wave at port 1 and 2 respectively. S_{21} and S_{12} are the transmission coefficient from port 1 to 2 and port 2 to 1 respectively. For a passive device that does not require external power, its reflection and transmission coefficient will not be greater than its incident wave. For filter, it has a reciprocal characteristic which means its two ports can be interchangeable because $S_{21} = S_{12}$.

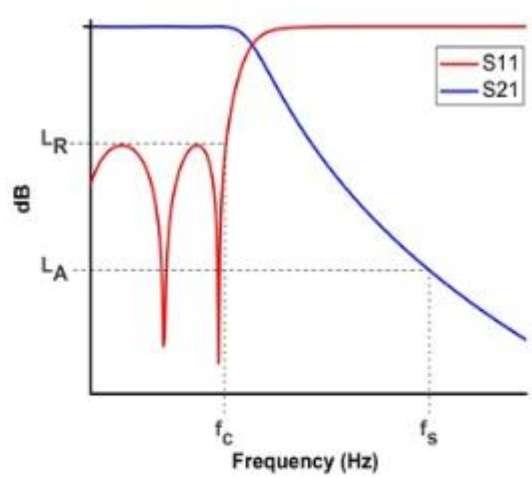


Figure 6: Filter response

When S_{12} and S_{11} are expressed in decibels, it is called insertion loss, L_A and return loss, L_R respectively. Insertion loss is a measure of attenuation through the two port network. A perfectly matched lossless network would have zero insertion loss but infinite return loss. Typical ‘good value’ for return loss is for a well matched system is between 15 to 25dB depends on the application.

2.4 Ladder Network

Ladder network is a network that composed of resistor, inductor or capacitor connected in cascaded T and pi network. It can be start with a series or shunt component first.

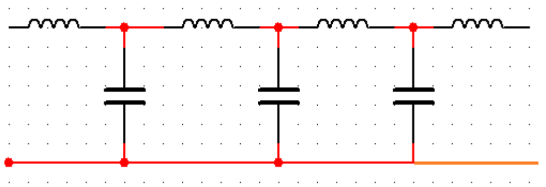


Figure 7: Series first

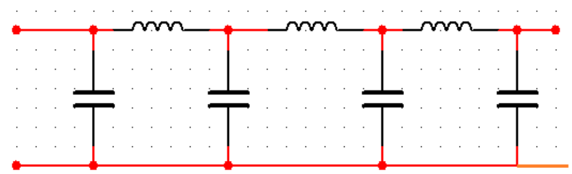


Figure 8: Shunt first

For a LC ladder network, it can be seen that the number of inductor used in series first network is more than shunt first network if the number of components used are the same. The number of component used is determined by its order. Ladder network is used because its response is least sensitive to the change in component value.

LC filter is selected because it can converts the discontinuous pulse width modulation (PWM) from Class D amplifier into a smooth analog sinusoid. Hence, LC filter can extracts audio signal from PWM and become audio signal. This filtering process is important because it reduces electromagnetic interference (EMI). The PWM contains high amplitude voltage signal and filter these PWM will also filter the high frequency content associated with the PWM, reducing EMI emissions. Besides, LC filter also reduce Class D amplifier ripple current. Class D amplifier with AD modulation which consist of two levels of PWM, there will be a ripple current superimposed on the audio signal. When there is LC filter, specifically as the cut off frequency of the LC filter reduced relative to PWM switching frequency of the amplifier, the ripple current will also reduce and only a small amount of ripple will be present. The reason class D amplifier is a concern because it is becoming popular in audio devices. This is because it has a high efficiency and is useful especially for portable battery driven products.

2.5 Types of filter

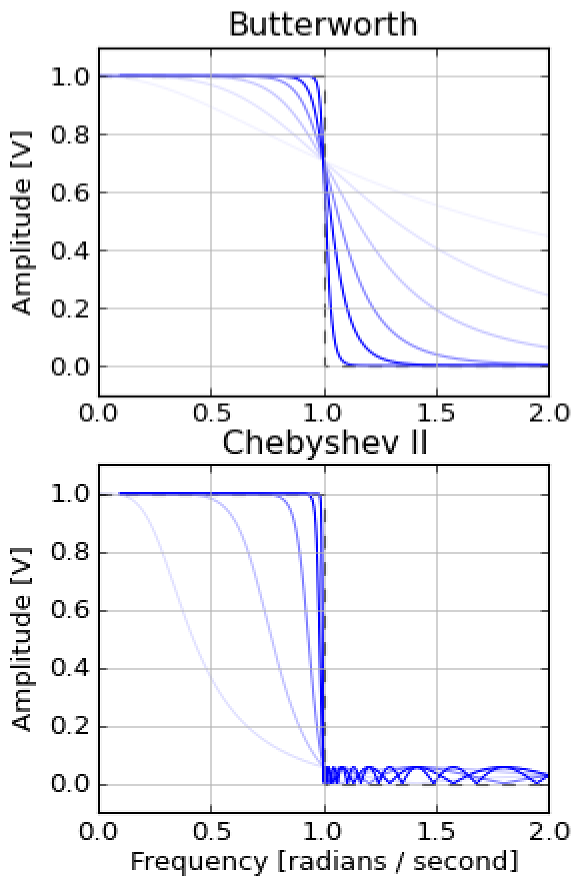


Figure 9: Types of filter response

There are few types of filters namely Butterworth, Chebyshev and Elliptic. Butterwoth has a flat response in passband and stopband. For Chebyshev Type 1 and 2, it will have ripple in the passband and stopband respectively. For elliptic, ripple will be present in both passband and stopband. The roll off rate increase when the number of order increase. Elliptic has the fastest roll off rate while Butterworth has the slowest roll off rate. In this project, we will use Chebyshev design because it has a higher roll off rate. Elliptic is not used because it has a response of transmission zeros are not located in infinity and in certain frequencies without flexibility in their location. Butterworth also can achieve a high roll off rate with higher filter order but this will increase the number of components. This is undesirable because the number of inductor will increase and the probability of distortion will increase too.

Chapter 3: Methodology

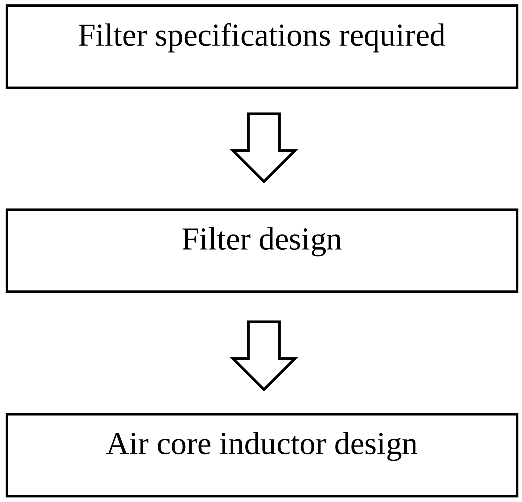


Figure 10: Flow Chart

First of all, the filter specifications needed to design is obtained. Next, filter is design based on the specifications. Some design parameters like type of filter, filter order, stability and component values are determined. Lastly, air core inductor is designed based on the filter design.

Chapter 4: Results and Discussion

4.1 Filter Design

The audio analyzer that we are working on is U8903B from Keysight. This model audio analyzer has a bandwidth of 10Hz to 96 kHz or 1.5 MHz depending on the installed option. Hence, two low pass filters are required for this bandwidth. A cutoff frequency of 100 kHz is needed for the audio analyzer bandwidth of 96 kHz. On the other hand, a 300 kHz is needed for the 1.5 MHz bandwidth. The audio analyzer has a specification to measure up to the fifth harmonic distortion. Hence, in order for the fifth harmonic to be 1.5 MHz or below, the maximum fundamental signal need to be 300 kHz or below. Moreover, the 1.5 MHz bandwidth option is ideal to analyze class D amplifier spectrum or switching supplies where frequency components or noise above the audio band will degrade the audio quality. The switching frequency of class D amplifier, sigma-delta converter and other modern audio devices regularly center around 250 kHz to 300 kHz. Noise that is being reflected into the audible range will directly detrimental the sound. Noise that is present above the audible range will detrimental the sound by causing problems like oscillation, overheating and damage components, loss of headroom and interference with connected to adjacent equipment. The 1.5 MHz bandwidth is to analyze the harmonics of this out of band noise.

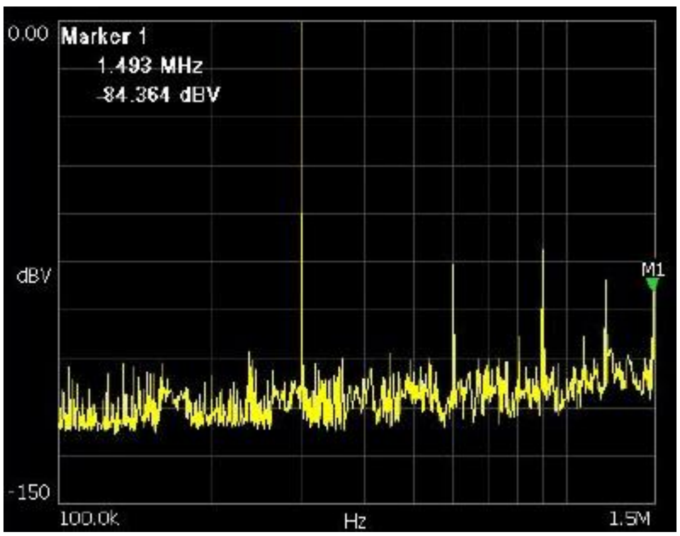


Figure 11: 300 kHz source and U8903B audio analyzer is able to measure up to fifth harmonics

The audio analyzer also comes with 24 bit resolution and two million point FFT. The higher the resolution, the better the level of detail can be seen above the noise floor. With a high bandwidth, 24 bit resolution permits low level out of band artifacts to be seen clearly in the presence of mush stronger fundamental.

To construct the low pass filter, we choose Chebyshev because it has the steeper roll off compare to Butterworth but has a lower roll off compare to elliptic. However, elliptic filter is harder to design. On the other hand, ripple will present in passband or stopband depends on the types of the Chebyshev filter. For type 1 Chebyshev filter, ripple will be present in passband while ripple will be present in type 2 Chebyshev filter.

The given specifications for the filter are as follows:

Table 2: Filter specifications

Cutoff frequency, f_c (kHz)	100	300
Stopband frequency, f_s (kHz)	200	600
Stopband attenuation, L_A (dB)	35	35
Passband return loss, L_R (dB)	15	15
Insertion loss, IL (dB)	1	1

The insertion loss at ripple level is expressed as

$$IL = 10 \log (1 + \varepsilon^2)$$

$$1 = 10 \log (1 + \varepsilon^2)$$

$$\varepsilon = 0.5088$$

However, ε will also affect the return loss value. The formula is given by

$$L_R = 10 \log [1/|S_{11}|^2]$$

$$15 = 10 \log [1/|S_{11}|^2]$$

$$|S_{11}|^2 = 0.0316$$

$$|S_{11}|^2 + |S_{12}|^2 = 1$$

$$|S_{12}|^2 = 1 - |S_{11}|^2$$

$$|S_{12}|^2 = 0.9684$$

$$IL = 10 \log [1/|S_{12}|^2] = 10 \log (1 + \varepsilon^2)$$

$$\varepsilon = 0.18 \text{ with } IL = 0.14$$

So, ε need to be lower than 0.5088 and below 0.18. Hence, $\varepsilon = 0.1$ is chosen.

S_{11} and S_{12} are the S parameters which are reflection coefficient and transmission coefficient respectively.

To determine which order of filter is used, this formula is used

$$N \geq (L_A + L_R + 6) / (20 \log_{10} [S + (S^2 -1)^{0.5}])$$

N = filter order

S = ratio of stopband to passband frequency

The order for both filters will be the same because its L_A, L_R and S are the same.

$$N \geq (35 + 15 + 6) / (20 \log_{10} [2 + (2^2 -1)^{0.5}])$$

$$N \geq 4.896$$

$$N = 5$$

Hence, fifth order filter is used and it is verified by Filpal software. The number of filter order is equivalent to the total number of inductor and capacitor in the circuit. A higher filter order will has a steeper roll off.

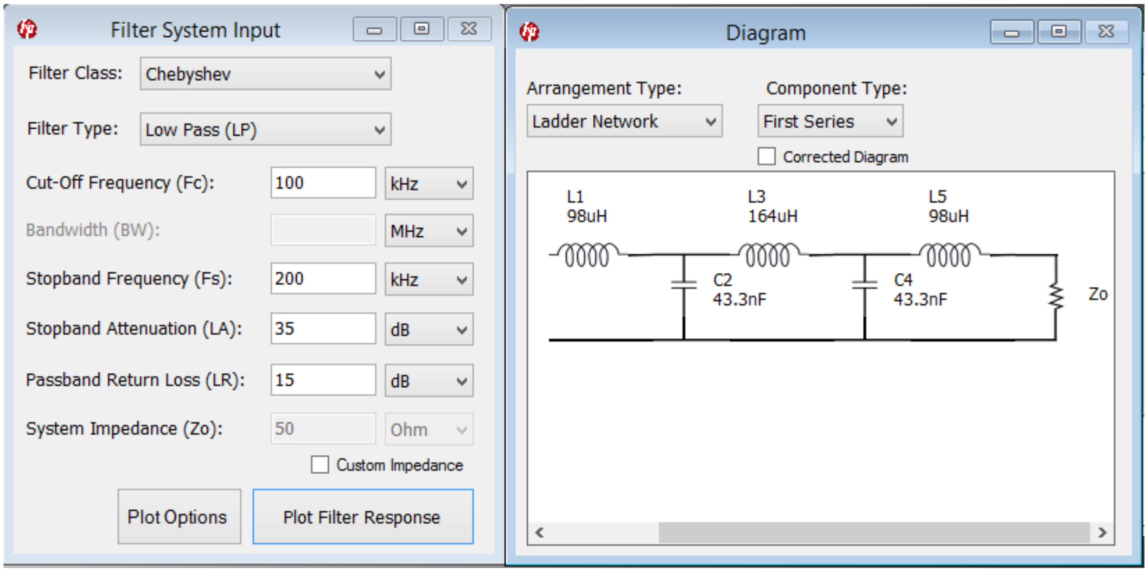


Figure 12: Filter design using Filpal

The Chebyshev polynomial is given by

$$T_{N+1}(\omega) = 2\omega T_N(\omega) – T_{N-1}(\omega)$$

with initial conditions $T_0(\omega) = 1$ and $T_1(\omega) = \omega$

So with these formulae, the fifth order of Chebyshev polynomial will be

$$T_5 = 16\omega^5 - 20\omega^3 + 5\omega$$

$|S_{12}(j\omega)|^2 = 1/[1+\varepsilon^2 T_N^2(\omega)]$

$|S_{12}(j\omega)|^2 = 1/ [1+0.01(-16js^5-20js^3-5js)^2]$ with $s = j\omega$

A system is said to be stable if all the components in the response from a finite set of initial conditions decay to zero as time increases. If the pole is laid in the positive real part of complex s-plane, the system is unstable and vice versa.

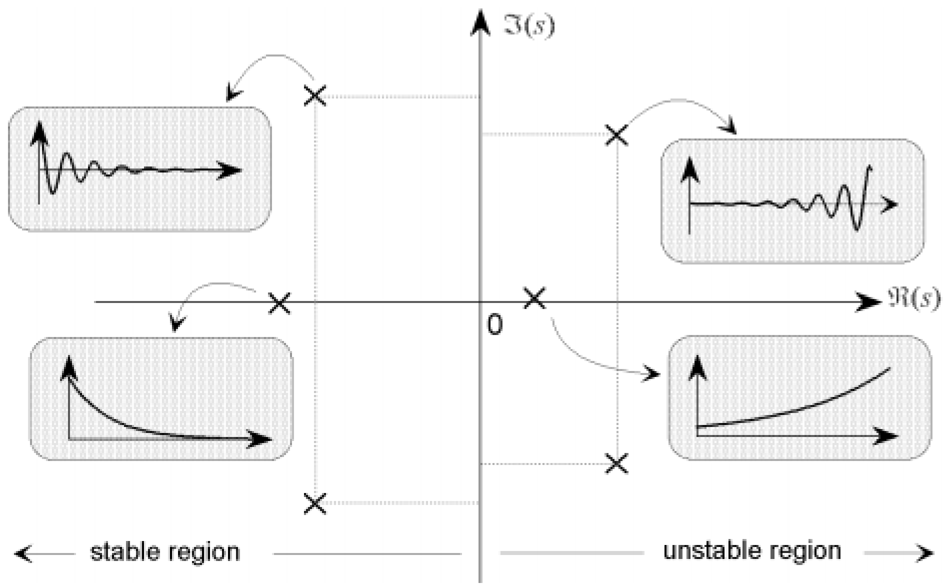


Figure 13: Analysis of system stability

The poles of $|S_{12}(j\omega)|^2$ are determined and only the one that are in the negative real part of complex s-plane are chosen.

$1+0.01(-16js^5-20js^3-5js)^2 = 0$

(s - 0.5147 - 0.6967j) (s + 0.1966 – 1.1272j) (s + 0.6362) (s + 0.1966 + 1.1272) (s - 0.5147 + 0.6967j) (s - 0.6362) (s - 0.1966 – 1.1272j) (s + 0.5147 – 0.6967j) (s + 0.5147 + 0.6967) (s - 0.1966 + 1.1272) =0

The poles that laid on the negative real part are

(s + 0.1966 – 1.1272j) (s + 0.6362) (s + 0.1966 + 1.1272) (s + 0.5147 – 0.6967j) (s + 0.5147 + 0.6967)

Therefore, the stable $|S_{12}(j\omega)|^2$ would be

$$|S_{12}(j\omega)|^2 = 1 / [(s + 0.1966 - 1.1272j)(s + 0.6362)(s + 0.1966 + 1.1272)(s + 0.5147 - 0.6967j)(s + 0.5147 + 0.6967)]^2$$

$|S_{12}(j\omega)|^2 = 1 / [(s + 0.1471 - 1.1032j)^2 (s + 0.5927 - 0.3421j)^2 (s + 0.4068 + 0.6369j)^2 (s + 0.4154 - 0.8136j)^2 (s + 0.5950 + 0.1723j)^2 (s + 0.1363 + 0.9497j)^2]$

However, the frequency response of $|S_{12}(j\omega)|^2$ is unacceptable because it shows the output of the transmission is more than the input.

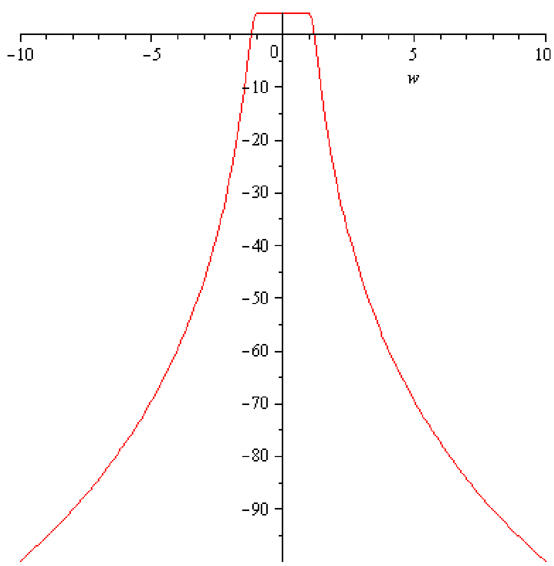


Figure 14: Response of $|S_{12}(j\omega)|^2$

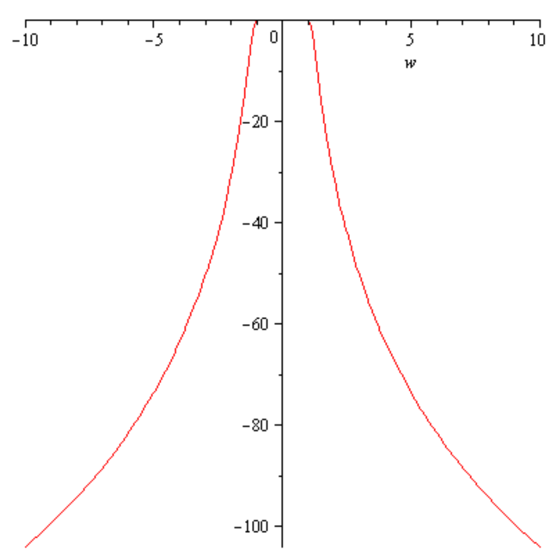


Figure 15: Shifted response of $|S_{12}(j\omega)|^2$

Hence, the response is limited by shifting the response downwards. So, the acceptable $|S_{12}(j\omega)|^2$ will be

$$|S_{12}(j\omega)|^2 = 0.3906\, |1/[(s + 0.1966 - 1.1272)(s + 0.6362)(s + 0.1966 + 1.1272)(s + 0.5147 - 0.6966)(s + 0.5147 + 0.6967)]|^2$$

A lowpass filter network can be synthesis by formulating $Y_{in}(p)$.

$$Y_{in}(p) = [1+S_{11}(p)] / [1-S_{11}(p)]$$

$$S_{11}(p) = \prod_{r=1}^{N} [p+j\cos\Theta_r] / [p+j\cos[\sin^{-1}(j\eta)+ \Theta_r]] \,, \qquad \text{r is the order of filter}$$

provided $j\cos[\sin^{-1}(j\eta)+ \Theta_r = \eta\sin\Theta_r+j(1+\eta^2)^{1/2}\cos\Theta_r$

$$\eta=\sinh\left(\frac{1}{N}\sinh^{-1}\frac{1}{\varepsilon}\right)$$

$$\Theta_r = \frac{(2r-1)\pi}{2N}$$

Hence, $S_{11}(p)=[p^5 + 0.3125p + 1.25p^3] / [2.0279p + 0.6252 + 3.3696p^3 + 3.2108p^2 + p^5 + 2.0588p^4]$

$$\begin{aligned}
Y_{in}(p) &= [1+S_{11}(p)] / [1-S_{11}(p)] \\
&= \left[1+ \frac{S11\,num}{S11\,den}\right] / \left[1- \frac{S11\,num}{S11\,den}\right] \\
&= \left[\frac{S11\,den+ S11\,num}{S11\,den}\right] / \left[\frac{S11\,den- S11\,num}{S11\,den}\right] \\
&= \frac{S11\,den+ S11\,num}{S11\,den - S11\,num}
\end{aligned}$$

$$\begin{aligned}
Y_{in}(p) &= 2.0279p +0.6252 +3.3696p^3 +3.2108p^2 +p^5 +2.0588p^4 + (p^5 +0.3125p +1.25p^3) / \\
&\quad 2.0279p +0.6252 +3.3696p^3 +3.2108p^2 +p^5 +2.0588p^4 - (p^5 +0.3125p +1.25p^3) \\
&= (2.3406p + 0.6252 + 4.6198p^3 +3.2108p^2 + 2p^5 + 2.0588p^4) / \\
&\quad (1.7152p + 0.6252 + 2.1195p^3 + 3.2108p^2 + 2.0588p^4)
\end{aligned}$$

The denominator of $Y_{in}(p)$ is always one power lower than its numerator.

A common realization of impedance and admittance function use in filter design is ladder network. It can be synthesized by continued fraction expansion.

It can be seen that $Y_{in}(p)$ tends to 0.9714p as p tends to infinity. Hence, the residue is evaluate at p = ∞.

$$\frac{Y_{in}(p)}{p}\Big|_{p=\infty} = 0.9714$$

A parallel capacitor of value C_1 = 0.9714 is removed, leaving admittance $Y_1(p)$ where

$$Y_1(p) = Y_{in}(p) - 0.9714p$$

$$= \frac{-1.7332p-0.6252-1.501p^3-1.5447p^2}{-0.6252-3.211p^2-2.0589p^4-1.7152p-2.1195p^3}$$

Now we invert $Y_1(p)$ to form impedance $Z_1(p)$ where

$$Z_1(p) = \frac{-0.6252-3.211p^2-2.0589p^4-1.7152p-2.1195p^3}{-1.7332p-0.6252-1.501p^3-1.5447p^2}$$

Again the residue at p = ∞ is evaluate

$$\frac{Z_1(p)}{p}\Big|_{p=\infty} = 1.3719$$

A series inductor value of L_2 = 1.3719 is removed leaving impedance Z_2 where

$$Z_2(p) = Z_1 - 1.3719p$$

$$= \frac{0.6252+0.8330p^2+0.8574p}{1.7332p+0.6252+1.5007p^3+1.5447p^2}$$

$$Y_2(p) = \frac{1.7332p+0.6252+1.5007p^3+1.5447p^2}{0.6252+0.8330p^2+0.8574p}$$

The residue at p = ∞ is evaluate

$$\frac{Y_2(p)}{p}\Big|_{p=\infty} = 1.8016$$

A parallel capacitor of value C_3 = 1.8016 is removed.

$$Y_3(p) = Y_2(p) - 1.8016p$$

$$= \frac{-0.6069p-0.6252}{-0.6252-0.8330p^2-0.8575p}$$

$Z_3(p) \quad = \frac{-0.6252-0.8330p^2-0.8575p}{-0.6069p-0.6252}$

$\frac{Z_3(p)}{p}\Big|_{p=\infty} \quad = 1.3726$

A series inductor value of L_4 = 1.3726 is removed.

$Z_4(p) \quad = Z_3(p) – 1.3726p$

$= \frac{0.6252}{0.6069p+0.6252}$

$Y_4(p) \quad = 0.9707p+1$

$\frac{Y_4(p)}{p}\Big|_{p=\infty} \quad = 0.9707$

A parallel capacitor of value C_5 = 0.9707 is removed.

Therefore,

$C_1 = 0.9714 \qquad L_2 = 1.3719$

$C_3 = 1.8016 \qquad L_4 = 1.3726$

$C_5 = 0.9707$

The above low pass filter synthesis is based on an angular cut off frequency, ω_c = 1rad/s operating from a 1Ω generator into a 1Ω load.

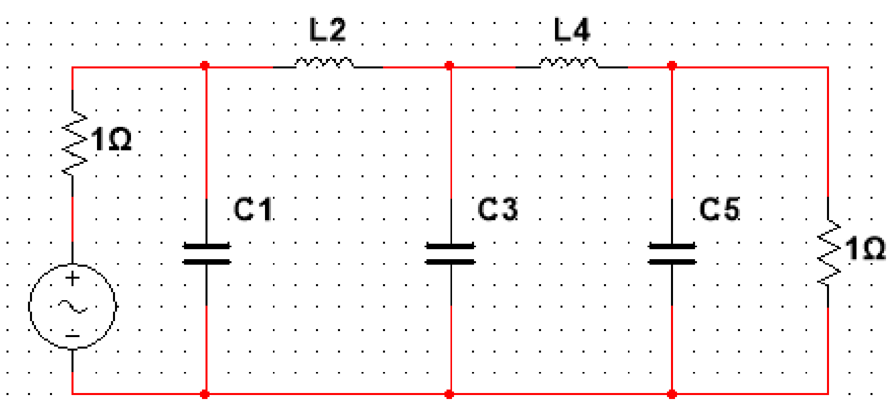


Figure 16: Synthesis of a ladder network

In reality, filter is designed based on arbitrary impedance level and arbitrary cutoff frequency. Most microwave filter operates in 50Ω impedance level to compromise the losses and power handling capacity of the coaxial cable. To convert the circuit elements in the filter to the required impedance level, the circuit elements are scaled by Z_0= 50Ω.

L => Z_0 L $\qquad\qquad$ C => $\frac{C}{Z_0}$

To convert the cut off frequency to the required ω_c , it is scaled with $\frac{1}{\omega_c}$

ω => $\frac{\omega}{\omega_c}$

The impedance for inductor X_L(jω) = j($\frac{\omega}{\omega_c}$)L , $\qquad$ that is L => $\frac{L}{\omega_c}$

For capacitor, X_c(jω) = $\frac{1}{j(\frac{\omega}{\omega_c})C}$, $\qquad$ that is that is C => $\frac{C}{\omega_c}$

Hence,

L => $\frac{Z_0 L}{\omega_c}$ $\qquad\qquad$ C => $\frac{C}{Z_0 \omega_c}$

For f_c = 100kHz

$C_1 = \frac{0.9714}{50(2\pi\times100000)}$ $= 30.9087\times10^{-9}$ F $= 30.9$nF

$L_2 = \frac{50(1.3719)}{(2\pi\times100000)}$ $= 109.12\times10^{-6}$ H $= 109$µH

$C_3 = \frac{1.8016}{50(2\pi\times100000)}$ $= 57.3224\times10^{-9}$ F $= 57.3$nF

$L_4 = \frac{50(1.3726)}{(2\pi\times100000)}$ $= 109.18\times10^{-6}$ H $= 109$µH

$C_5 = \frac{0.9707}{50(2\pi\times100000)}$ $= 30.8850\times10^{-9}$ F $= 30.9$nF

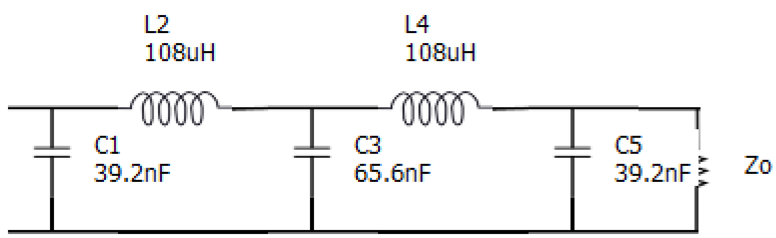


Figure 17: Value of circuit elements obtained from Filpal with f_c = 100kHz

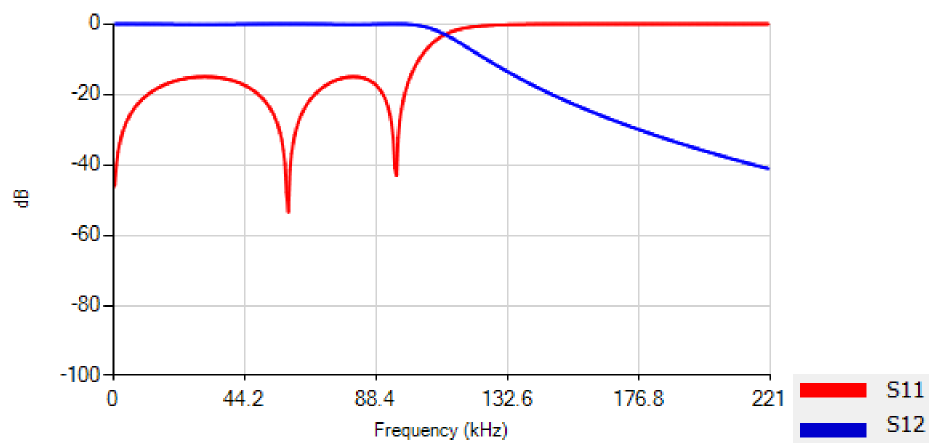


Figure 18: Filter response for the circuit shown in Figure 16

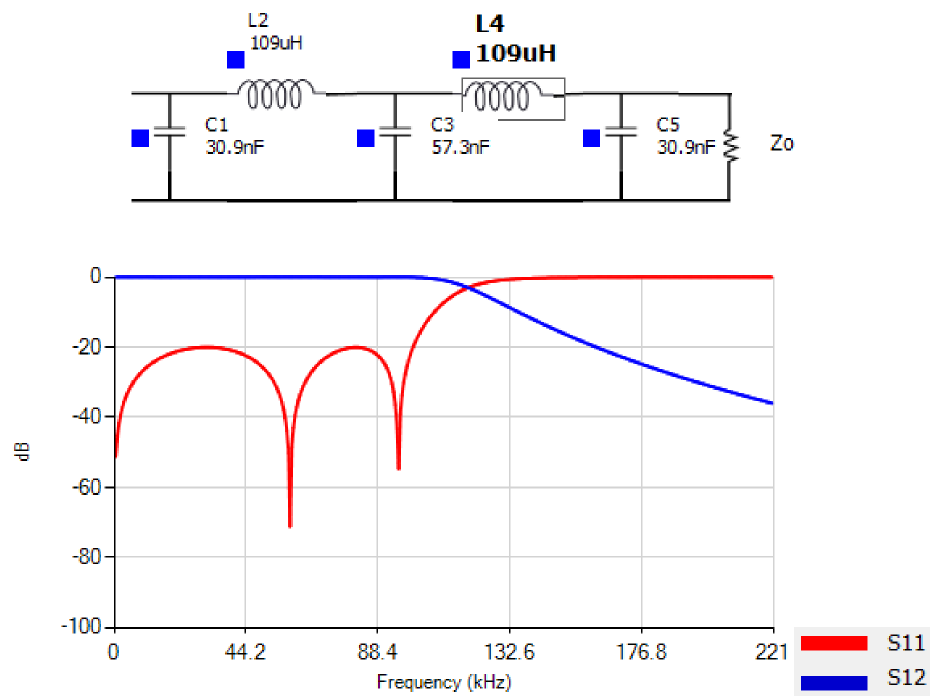


Figure 19: Filter response based on the calculated value for f_c = 100kHz

For another low pass filter with cutoff frequency of 300kHz, the same Y_{in} can be used because it is dependent to S_{11} and S_{11} is dependent to filter order, r and η which is dependent to filter order and ε which will affects the insertion loss, IL. The filter order is affected by stopband attenuation, L_A, passband return loss, L_R and the ratio of stopband to passband frequency, S. Since filter specifications corresponding to L_A, L_R, S and IL are similar for both filter, therefore their Y_{in} will be similar too.

For f_c = 300kHz

$C_1 = \frac{0.9714}{50(2\pi\times300000)}$ $= 10.3029\times10^{-9}$ F $= 10.3$nF

$L_2 = \frac{50(1.3719)}{(2\pi\times300000)}$ $= 36.3757\times10^{-6}$ H $= 36\mu$H

$C_3 = \frac{1.8016}{50(2\pi\times300000)}$ $= 19.1075\times10^{-9}$ F $= 19.1$nF

$L_4 = \frac{50(1.3726)}{(2\pi\times300000)}$ $= 36.3959\times10^{-6}$ H $= 36\mu$H

$C_5 = \frac{0.9707}{50(2\pi\times300000)}$ $= 10.2950\times10^{-9}$ F $= 10.3$nF

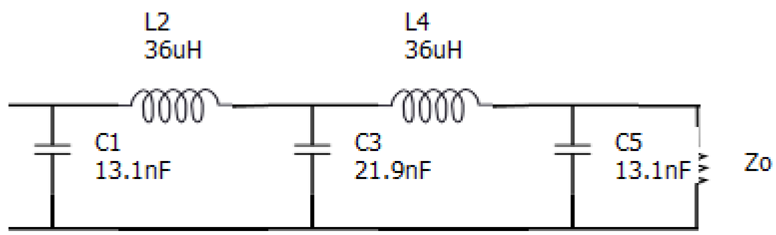


Figure 20: Value of circuit elements obtained from Filpal with f_c = 300kHz

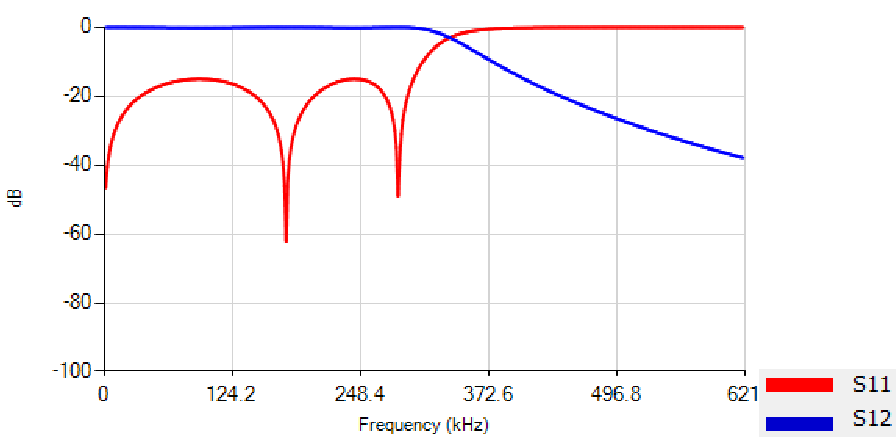


Figure 21: Filter response for the circuit shown in Figure 19

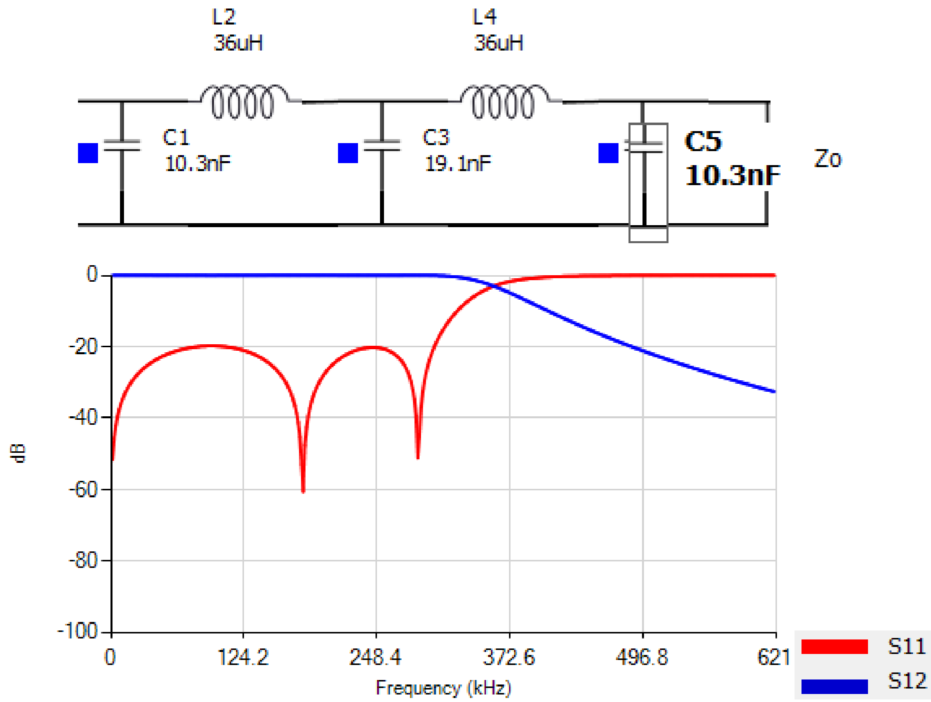


Figure 22: Filter response based on the calculated value for f_c = 300kHz

The results of filter response between simulated from the specifications given and calculated is almost similar just that the filter response from calculation has a lower L_R which is better. Hence, inductor value of 109µH and 36µH are needed to be construct. Inductor with air core is used because the inductance will not be affected by current where there is no ferromagnetic core to reach saturation point as the current increased.

4.2 Inductor Design

Table 3: Inductors design parameters

Inductance (μH)	110	36
Diameter of coil former (mm)	3	3
Length of winding (mm)	20.9	12.84
AWG	19	22
Diameter of wire without insulation (mm)	0.9117	0.6439
Diameter of wire with insulation (mm)	0.9937	0.7018
Number of turns	189	124
Numbers of layers	9	7
Winding thickness (mm)	8.944	4.9123
Required length of wire (m)	7.091	3.056
DC resistance of coil	0.19	0.16

These values are obtained through online multilayer coil inductance calculator. Input data are keyed in and it will calculate the other parameters needed.

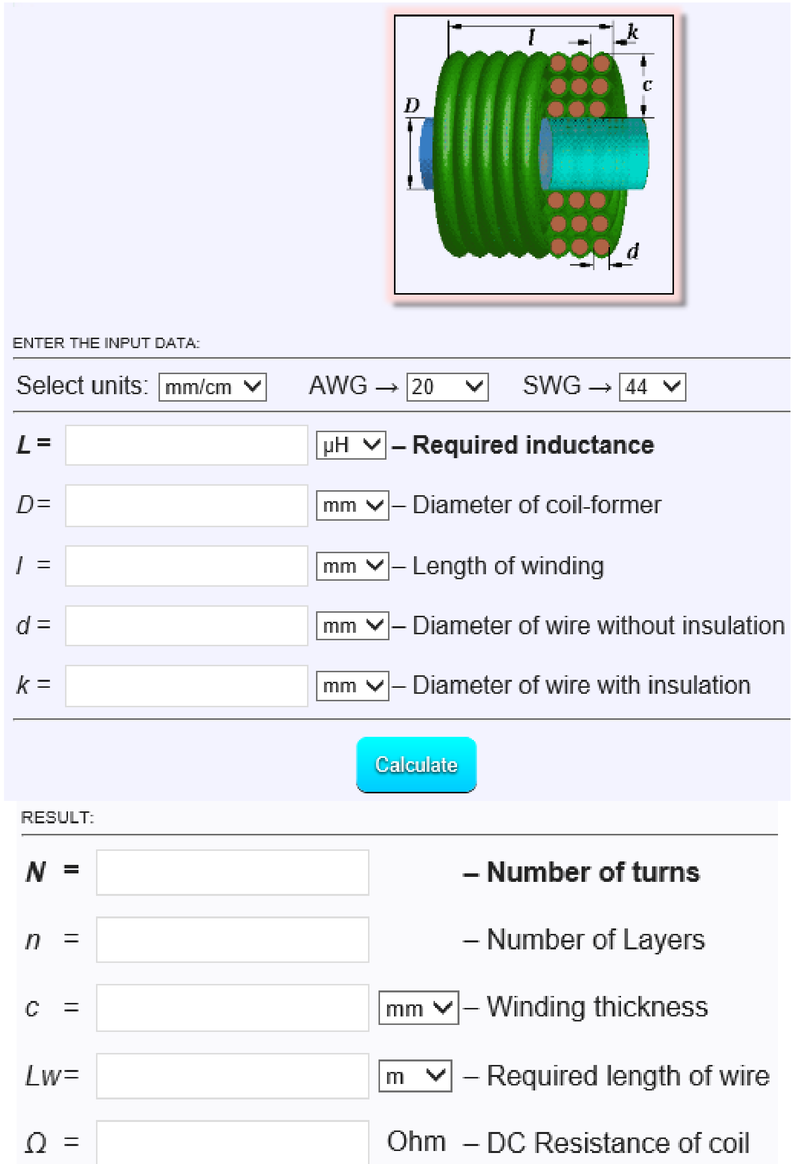


Figure 23: Online inductance calculator

Ways to determine inductor parameter

Firstly, the specifications of the inductor is determined. The inductance value needed are 36µH and 110µH with the DC resistance less than 0.2Ω. A low DC resistance is essential for low power consumption, low loss and high output. Next, a square shape inductor is desired to minimize the space of inductor consumed.

There are some of the design that meets the requirements of square shape with DC resistance less than 0.2Ω.

These are the inductor designs with 110µH at the same time meet the requirements using AWG 19. Height = D + 2c; dimension = l × height

Table 4: Parameters of AWG 19 with length 22.9mm for 110µH

l (mm)	22.9							
D (mm)	14	12	11	10	9	8	6	4
C (mm)	4.96	5.96	5.96	5.96	6.96	6.96	7.95	8.94
Lw (m)	6.57	6.707	6.67	6.723	6.83	6.84	7.08	7.24
Ω	0.18	0.18	0.18	0.18	0.18	0.18	0.19	0.19
Height(mm)	23.92	23.92	22.92	21.92	22.92	21.92	21.9	21.88
Dimension (mm × mm)	22.9 × 23.92	22.9 × 23.92	22.9 × 22.92	22.9 × 22.92	22.9 × 22.92	22.9 × 21.92	22.9 × 21.9	22.9 × 21.88

Table 5: Parameters of AWG 19 with length 21.9mm for 110µH

l (mm)	21.9						
D (mm)	8	6	5	4	3	2	1
c (mm)	6.96	7.95	7.95	8.94	8.94	9.94	9.94
Lw (m)	6.78	6.92	6,97	7.132	7.117	7.266	7.25
Ω	0.18	0.19	0.19	0.19	0.19	0.19	0.19
Height(mm)	21.92	21.9	20.9	21.88	20.88	21.88	20.88
Dimension (mm × mm)	21.9 × 21.92	21.9 × 21	21.9 × 20.9	21.9 × 21.88	21.9 × 20.88	21.9 × 21.88	21.9 × 20.88

Table 6: Parameters of AWG 19 with length 20.9mm for 110µH

l (mm)	20.9
D (mm)	3
c (mm)	8.94
Lw (m)	7.091
Ω	0.19
Height (mm)	20.88
Dimension (mm × mm)	20.9 × 20.88

For 110µH inductor, 20.9mm × 20.88mm with D = 3mm AWG 19 design is chosen because it has the smallest dimension.

For 36µH inductor, these are the available designs that meets the requirements.

Table 7: Parameters of AWG 19 with length 19mm for 36µH

AWG	19			
l (mm)	18.9	17.9	16.9	15.9
D (mm)	11	8	5	3
c (mm)	3.97	4.97	5.96	6.96
Lw (m)	3.35	3.40	3.46	3.50
Ω	0.09	0.09	0.09	0.09
Height (mm)	19	18	17	17
Dimension (mm × mm)	18.9 × 19	17.9 × 18	16.9 × 17	15.9 × 17

Table 8: Parameters of AWG 22 and 23 for 36µH

AWG	22	23	
l (mm)	12.64	14.38	13.75
D (mm)	3	8	8
c (mm)	4.92	2.5	2.5
Lw (m)	3.056	2.84	2.83
Ω	0.16	0.19	0.19
Height (mm)	12.84	13	13
Dimension (mm × mm)	12.64 × 12.84	13 × 13	13.75 × 13

For 36µH inductor, 12.64mm × 12.84mm is used. It is compared with AWG 19 which is used for 110µH inductor. AWG 19 has a bigger dimension. Hence 12.64mm × 12.84mm with D = 3mm and 0.16Ω is chosen.

Chapter 5: Conclusion

In conclusion, a passive Chebyshev ladder network is designed to meet the requirements of the audio analyzer.

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