Ultra Low Frequency Crossover Filter

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ABSTRACT

Our group designed a crossover filter, a circuit that separates input frequency into two separate frequency ranges for the output. The 2-way crossover filter will be able to take an audio signal input and split it into two output frequency ranges. One signal, filtered from 5Hz to 1 kHz, will connect to a woofer, while the other signal, filtered from 1 kHz to 20 kHz, will connect to a tweeter.

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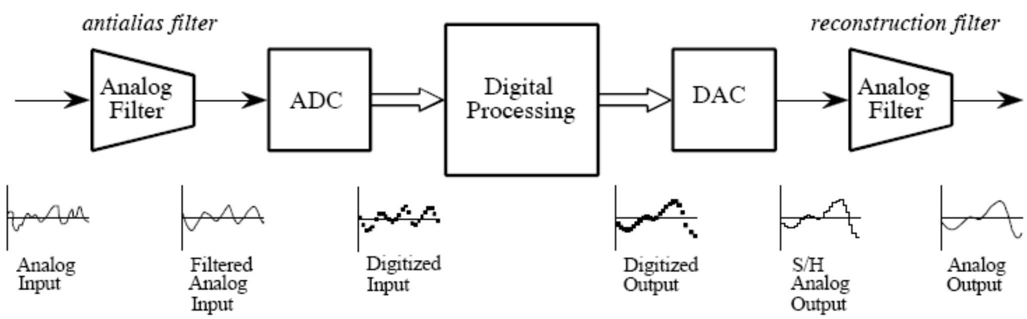
1. INTRODUCTION

1.1 Purpose and Motivation

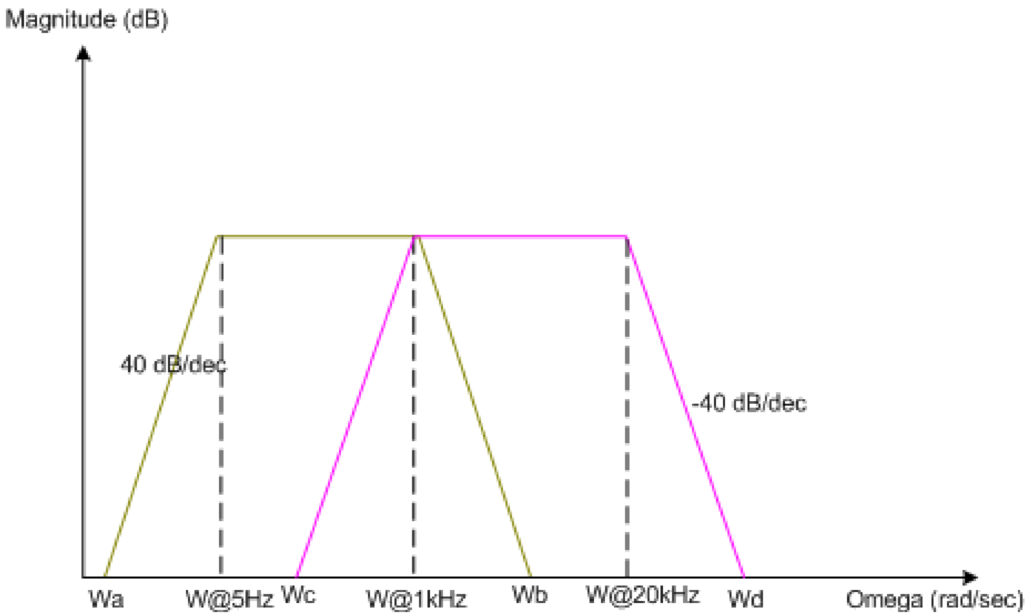
Digital Signal Processor (DSP) is widely used in today's applications. The main advantage of the DSP is its flexibility. Usage of digital filters has been a growing trend, and is gradually replacing analog filters, although a solid understanding of analog filters is still required in order to control the DSP system. Designing a DSP without knowing the properties of analog filters, such as anti-alias and reconstruction filters, will result in unsuccessful software implementations of the DSP. For example, the characteristics of digitized input signal is dependant on what type of anti-alias filter used at the input. Software tricks can replace the need of the anti-alias and reconstruction filters, but naturally they require an understanding of analog hardware versions. The main goal of the DSP and other digital filters is to replace hardware with software. Such embedded software designs will decrease the costs and increase the flexibility of circuits. Many DSP applications are related to digital filter design. [2]

Fig.1: Analog filter and DSP relationship. [2]



1.2 Specifications

The crossover filter has crossover frequency at 1 kHz.
A slope of at least 40dB/dec is preferred for sharper attenuation
5 Hz - 20 kHz frequency band



1.3 Optional Projects

The analog crossover filter can be compared with that of digital filter. A Texas Instruments TMS320VC5510 DSP Kit will be used to design and evaluate a digital filter, and compared with the implementation process and performance of an analog filter.

2. DESIGN APPROACH

2.1 Analog Filter Design Decisions

Since the design of the crossover filter was divided into band pass and crossovers filter, there were several options for each filter design. A Sallen-Key design has smaller band pass ripple, and is preferred over RLC circuit filter design due to audio quality degraded by such ripple. Another reason for selecting the Sallen-Key design choice is that it doesn't use inductors, which is useful as inductors can be costly and bulky. The LF351 op-amp was used for this filter design. The LF351 is JFET input operation amplifier with an internally compensated input offset voltage. The JFET input device provides wide bandwidth, low bias currents and offset currents. The values of the base capacitors and resistors were arbitrary selected at R = 10kohms, and C = 0.01uF.

The fixed-point DSP (Texas Instruments TMS320VC5510 DSP Kit) was the choice because it was on sale, was well documented, possessed built-in A/D converters, came with two stereo inputs and outputs, and cost $100 less than a floating- point DSP Kit. Using MATLAB's filter design tool to design Finite Impulse Response (FIR) with Blackman window method filter and exporting it into Code Composer Studio to program DSP. FIR filter was used due to fixed-point DSP limitation. Additionally the Blackman window method was selected since it is popular for audio application and it damps out the effect of truncation of infinite series or Gibbs phenomenon. [2]

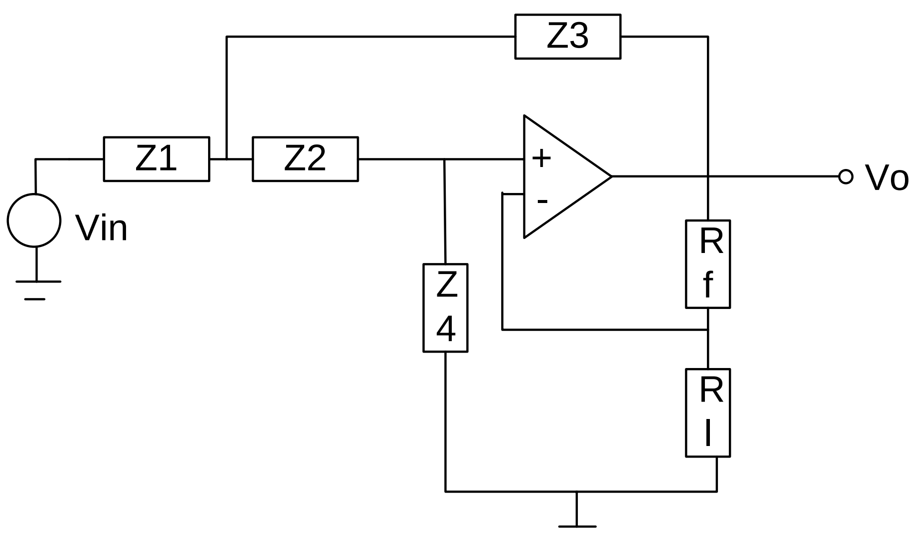
Commonly used analog filters are Chebyshev, Butterworth and Bessel filters. Each filter can be adjusted by selecting the number of poles and zeros. The more poles, the more electronic components it requires and the better it performs.

2.2 Tools and Equations

Implementation of the Bessel, Chebyshev, Butterworth filters is done utilizing Sallen Key design as below.

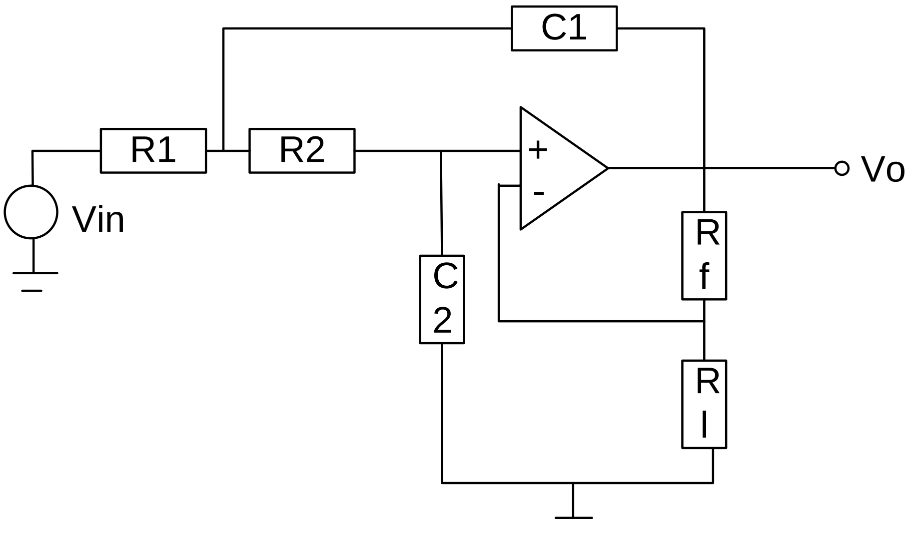
Transfer function for this topology is shown as below:

Fig.2.2: General Sallen Key Topology and Transfer Function



$$\frac{V_0}{V_{in}} = \frac{A_0 Z_3 Z_4}{Z_1 Z_2 + Z_2 Z_3 + Z_3 Z_4 + Z_1 Z_3 + Z_1 Z_4 (1 - A_0)}$$

Fig.2.2a: Sallen Key Low Pass Filter Transfer Function



$$H(s) = \frac{V_0}{V_{in}} = \frac{A_0 / C_1 C_2 R_1 R_2}{S^2 + [\frac{1-A_0}{C_2 R_2} + \frac{1}{C_1 R_2} + \frac{1}{C_1 R_1}]S + \frac{1}{C_1 C_2 R_1 R_2}}$$

Further simplify this equation by:

Letting $R_1 = R_2 = R$ and $\mathrm{C}_1 = C_2 = C$

critical frequency $\omega_0 = \frac{1}{RC}$

pass band gain $\mathrm{A}_0 = 1 + \frac{R_f}{R_l}$

Compare with Appendix 7 to obtain parameters shown in Table2.2a.

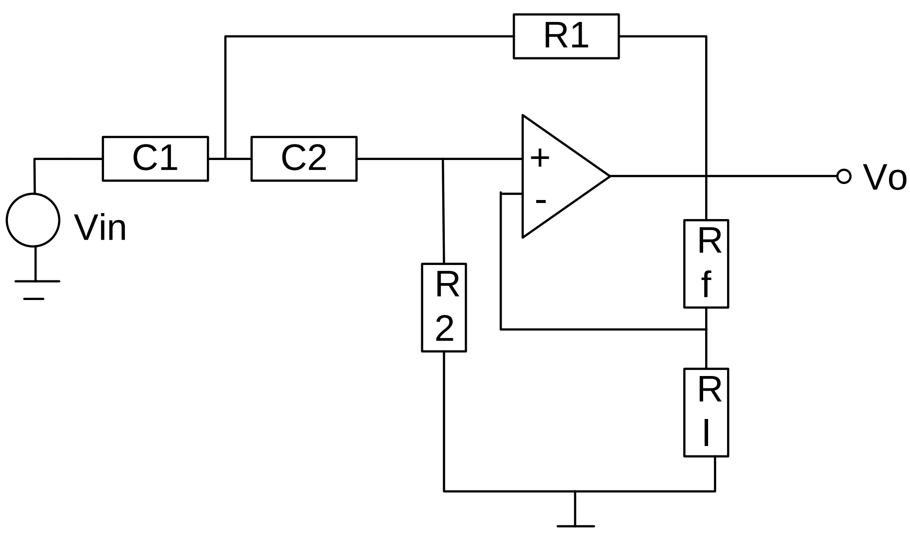
Table.2.2a: Filter Design Parameters [2]

Parameters for designing Bessel, Butterworth, and Chebyshev (6% ripple) filters.

# poles		Bessel k_1	Bessel k_2	Butterworth k_1	Butterworth k_2	Chebyshev k_1	Chebyshev k_2
2	stage 1	0.1251	0.268	0.1592	0.586	0.1293	0.842
4	stage 1	0.1111	0.084	0.1592	0.152	0.2666	0.582
	stage 2	0.0991	0.759	0.1592	1.235	0.1544	1.660
6	stage 1	0.0990	0.040	0.1592	0.068	0.4019	0.537
	stage 2	0.0941	0.364	0.1592	0.586	0.2072	1.448
	stage 3	0.0834	1.023	0.1592	1.483	0.1574	1.846
8	stage 1	0.0894	0.024	0.1592	0.038	0.5359	0.522
	stage 2	0.0867	0.213	0.1592	0.337	0.2657	1.379
	stage 3	0.0814	0.593	0.1592	0.889	0.1848	1.711
	stage 4	0.0726	1.184	0.1592	1.610	0.1582	1.913

To change from low pass to high pass filter, perform the swapping R and C.

Fig.2.2b: Sallen Key High Pass Filter Transfer Function



$$H(s)=\frac{V_0}{V_{in}}=\frac{A_0S^2}{S^2+[\frac{1-A_0}{C_1R_1}+\frac{1}{C_1R_2}+\frac{1}{C_2R_2}]S+\frac{1}{C_1C_2R_1R_2}}$$

2.3 Design Details

Step 1. Transfer function.
Step 2. Select a circuit that promises to be able to satisfy the required magnitude.

Deploying the Sallen Key topology as following.

Fig. 2.3a: Low Pass Filter with Sallen Key Topology

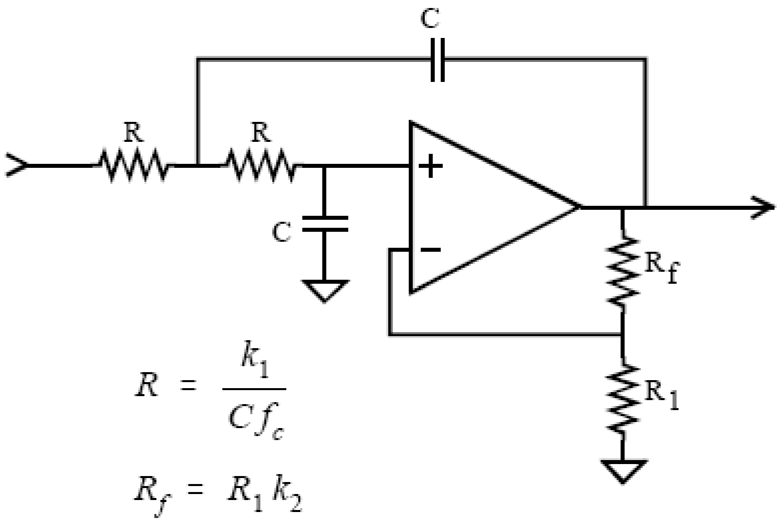
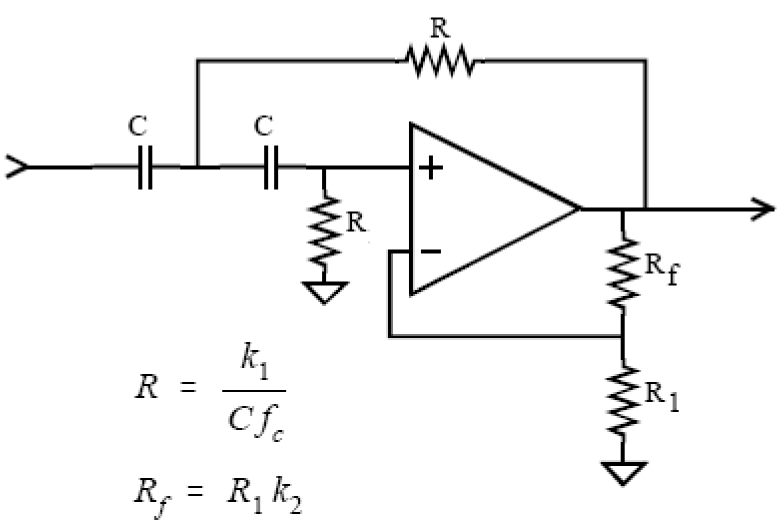


Fig.2.3b: High Pass Filter with Sallen Key Topology



Step 3. Determine the element values from the values of the poles and zeros.

Selecting a common values of capacitor for op-amp is C=0.01uF. Using the 0.01uF capacitor in the circuit and let's pick a resistor value as 10K ohm as typical value.

After PSPICE simulation with those values, Sallen-Key topology with Butterworth combination worked best. It is not important that which Op-amp is used because unity gain frequency is more than 30 times higher than the cutoff frequency and is less than 100 kHz. [2]

Now Design 2-stage crossover filter using Butterworth on the Sallen-Key scheme with C=0.01uF , R1=10k ohm, and cutoff frequency at 1000 Hz. The high-pass filter is just switched R and C version without changing the values of R_1 and R_f.

Butterworth filter parameters are
Stage 1: K1=0.1592, K2=0.152.
Stage 2: K1=0.1592, K2=1.235.

Calculated Results Table at fc = 5Hz, 1 kHz, and 20 kHz

Butterworth	**k1**	**k2**	**Actual Value Used**	
	R (ohm)	Rf (ohm)	R (ohm)	Rf (ohm)
Stage 1	3184000	1520	3000000	1500
Stage 2	3184000	12350	3000000	12300

Bessel	**k1**	**k2**
	R (ohm)	Rf (ohm)
Stage 1	2222000	840
Stage 2	1982000	7590

Chebyshev	**k1**	**k2**
	R (ohm)	Rf (ohm)
Stage 1	5332000	5820
Stage 2	3088000	16600

fc (Hz)	C (F)	R1 (ohm)
1000	0.00000001	10000

Butterworth	k1	k2	Actual Value Used	
	R (ohm)	Rf (ohm)	R (ohm)	Rf (ohm)
Stage 1	15920	1520	16000	1500
Stage 2	15920	12350	16000	12300

Bessel	k1	k2
	R (ohm)	Rf (ohm)
Stage 1	11110	840
Stage 2	9910	7590

Chebyshev	k1	k2
	R (ohm)	Rf (ohm)
Stage 1	26660	5820
Stage 2	15440	16600

fc (Hz)	C (F)	R1 (ohm)
20000	0.00000001	10000

Butterworth	k1	k2	Actual Value Used	
	R (ohm)	Rf (ohm)	R (ohm)	Rf (ohm)
Stage 1	796	1520	800	1500
Stage 2	796	12350	800	12300

Bessel	k1	k2
	R (ohm)	Rf (ohm)
Stage 1	555.5	840
Stage 2	495.5	7590

Chebyshev	k1	k2
	R (ohm)	Rf (ohm)
Stage 1	1333	5820
Stage 2	772	16600

Error Calculations:

fc (Hz)	C (µF)	R1 (ohm)
5	0.01	10000
1000	0.01	10000
20000	0.01	10000

Butterworth	k1	k2
	fc = 5Hz	
	R (ohm)	Rf (ohm)
Stage 1	3184000	1520
Stage 2	3184000	12350
	fc = 1kHz	
Stage 1	15920	1520
Stage 2	15920	12350
	fc = 20kHz	
Stage 1	796	1520
Stage 2	796	12350

Actual Value Used		at fc = 5 Hz	
R (ohm)	Rf (ohm)	% Error R	% Error Rf
3000000	1500	5.779%	1.316%
3000000	12300	5.779%	0.405%

At	5	Hz
Ave.%Er =		3.32%

Actual Value Used		at fc = 1k Hz	
R (ohm)	Rf (ohm)	% Error R	% Error Rf
16000	1500	0.503%	1.316%
16000	12300	0.503%	0.405%

At	1000	Hz
Ave.%Er =		0.68%

Actual Value Used		at fc = 20k Hz	
R (ohm)	Rf (ohm)	% Error R	% Error Rf
800	1500	0.503%	1.316%
800	12300	0.503%	0.405%

At	20000	Hz
Ave.%Er =		0.68%

	Ave%Error
at 5Hz	3.32
at 1kHz	0.68
at 20kHz	0.68

Total Ave.%Error	1.56	%

Total Avg. Error: 1.56%

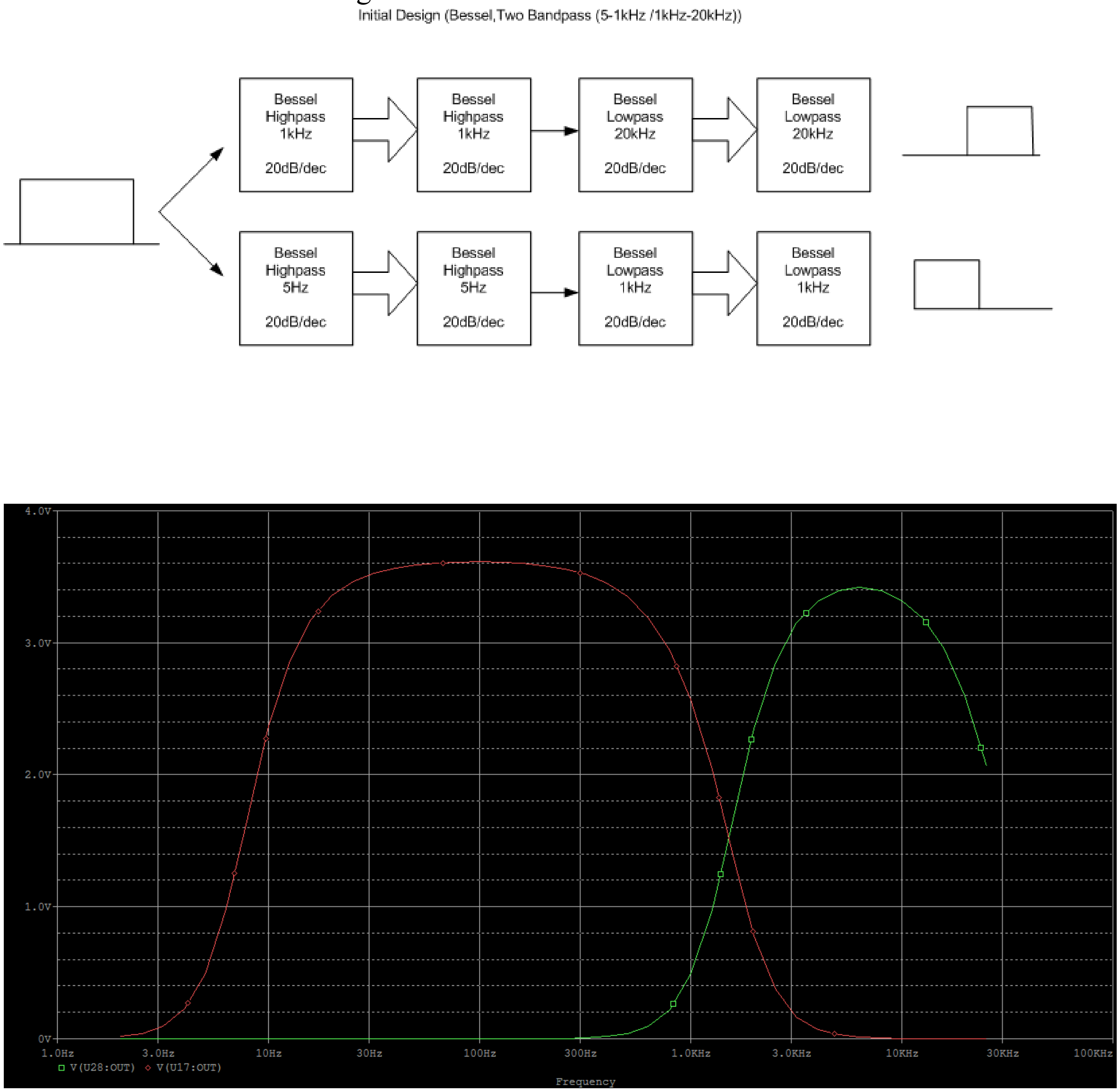
The average error calculated here is due to approximations in the purchasing of component parts. We could not find exact resistor values that we calculated, and thus chose the closest standard resistor values.

3. RESULTS

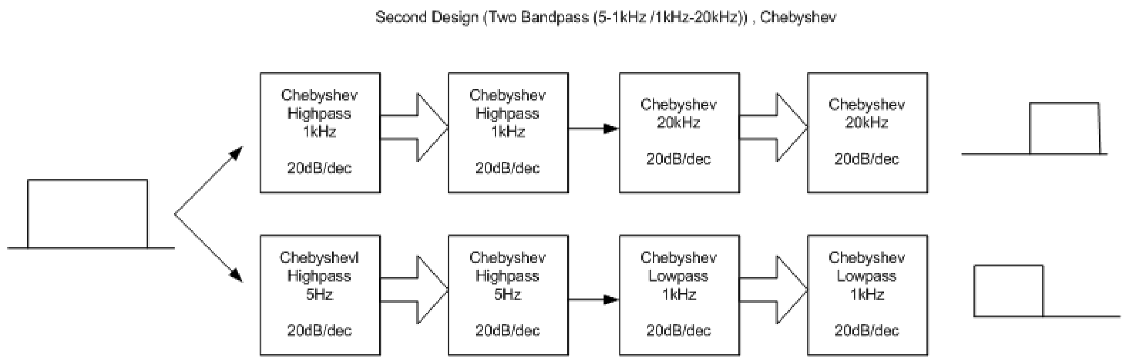
3.1. Simulations

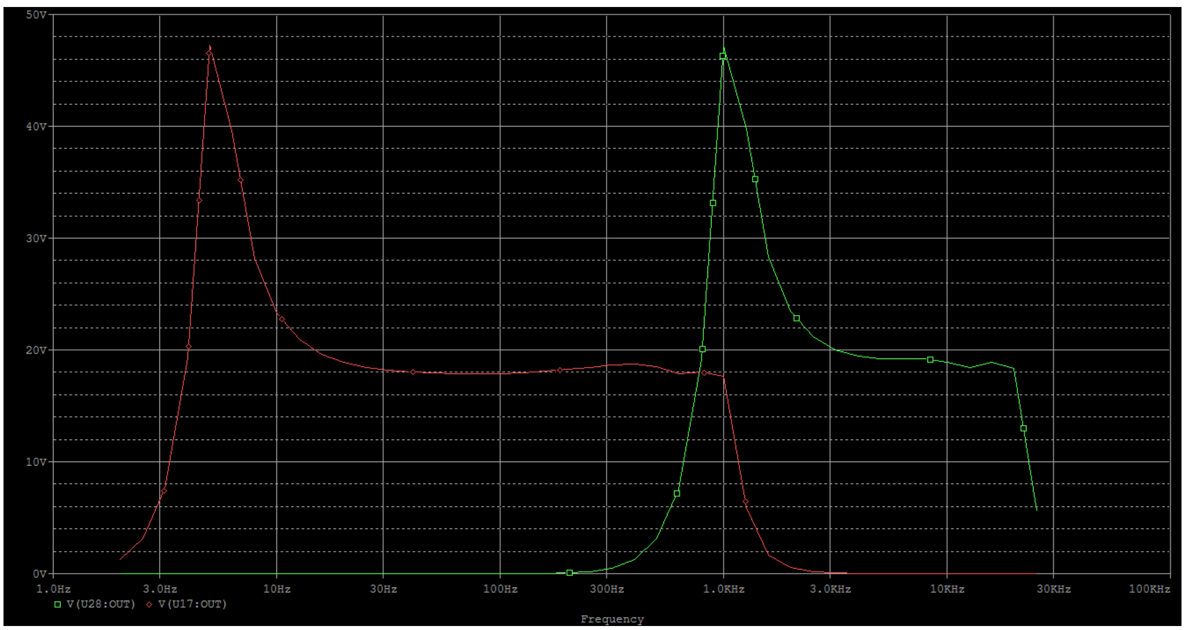
<u>Bessel Filter</u>

Fig.3.1a: Bessel Filter Block Diagram and PSPICE Simulation



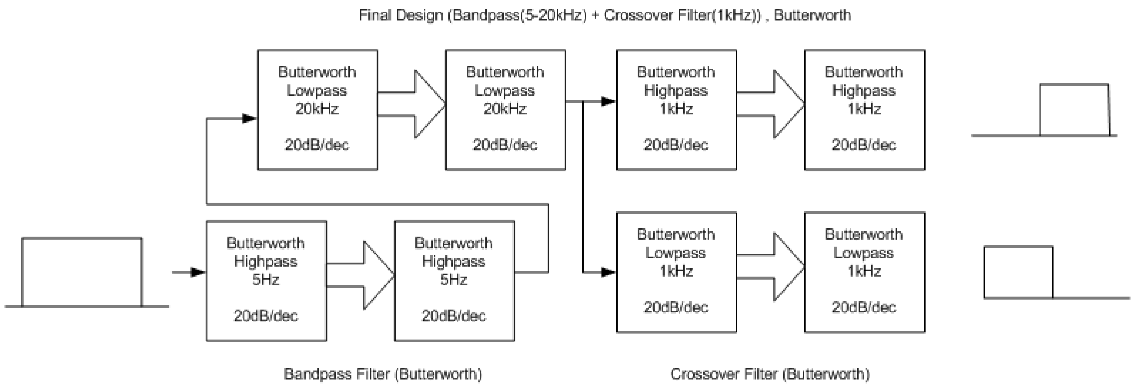
Chebyshev Filter
Fig.3.1b: Chebyshev Filter Block Diagram and PSPICE Simulation

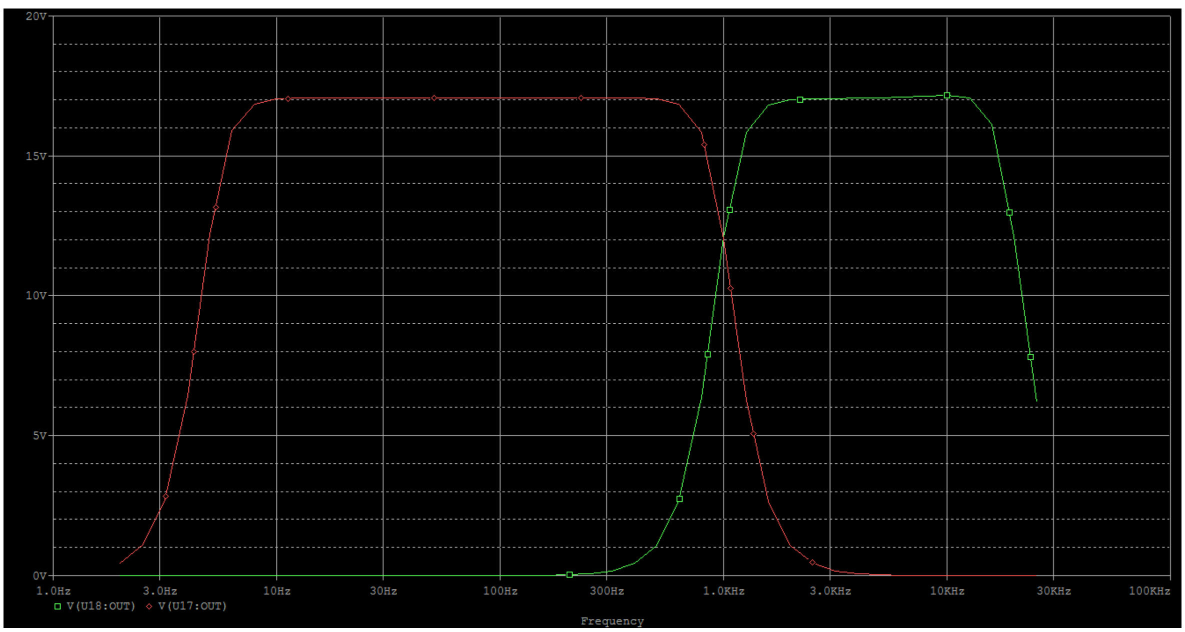




Butterworth Filter

Fig.3.1c: Butterworth Filter Block Diagram and PSPICE Simulation





Digital Filter

Using MATLAB filter design toolbox, fdatool command to design filter, simulate, and calculate coefficients for the DSP.

Fig.3.1d: MATLAB Simulation: 5-1kHz FIR Blackman window Band-pass filter Magnitude and Phase Plot

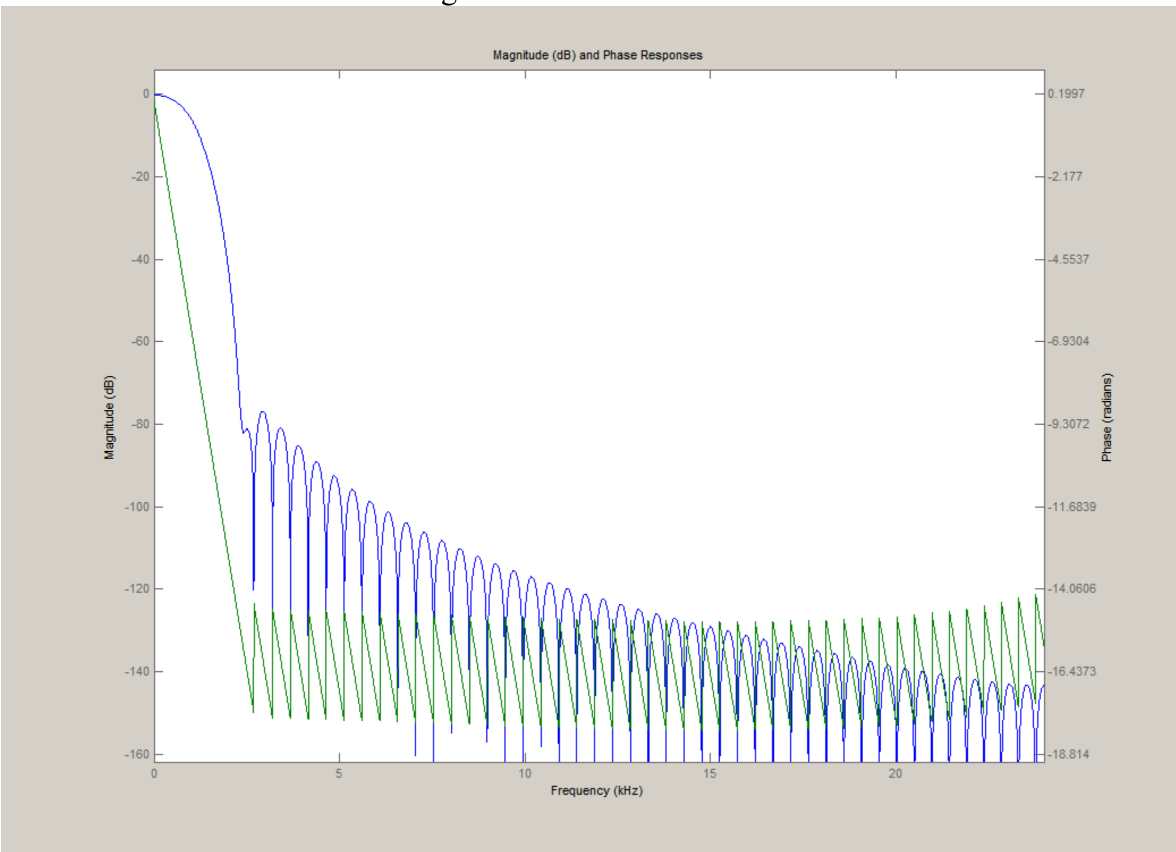


Fig.3.1e: Impulse Response Plot

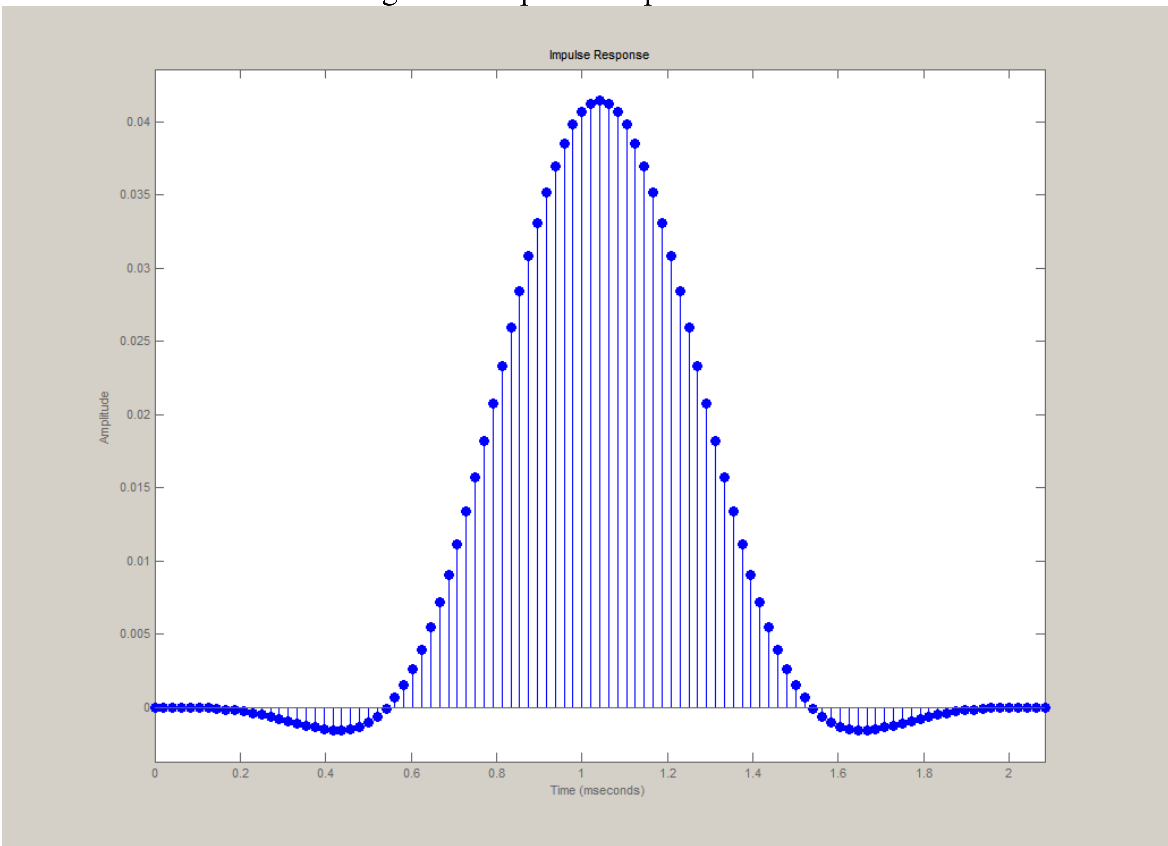


Fig.3.1f: Poles/Zeros Plot

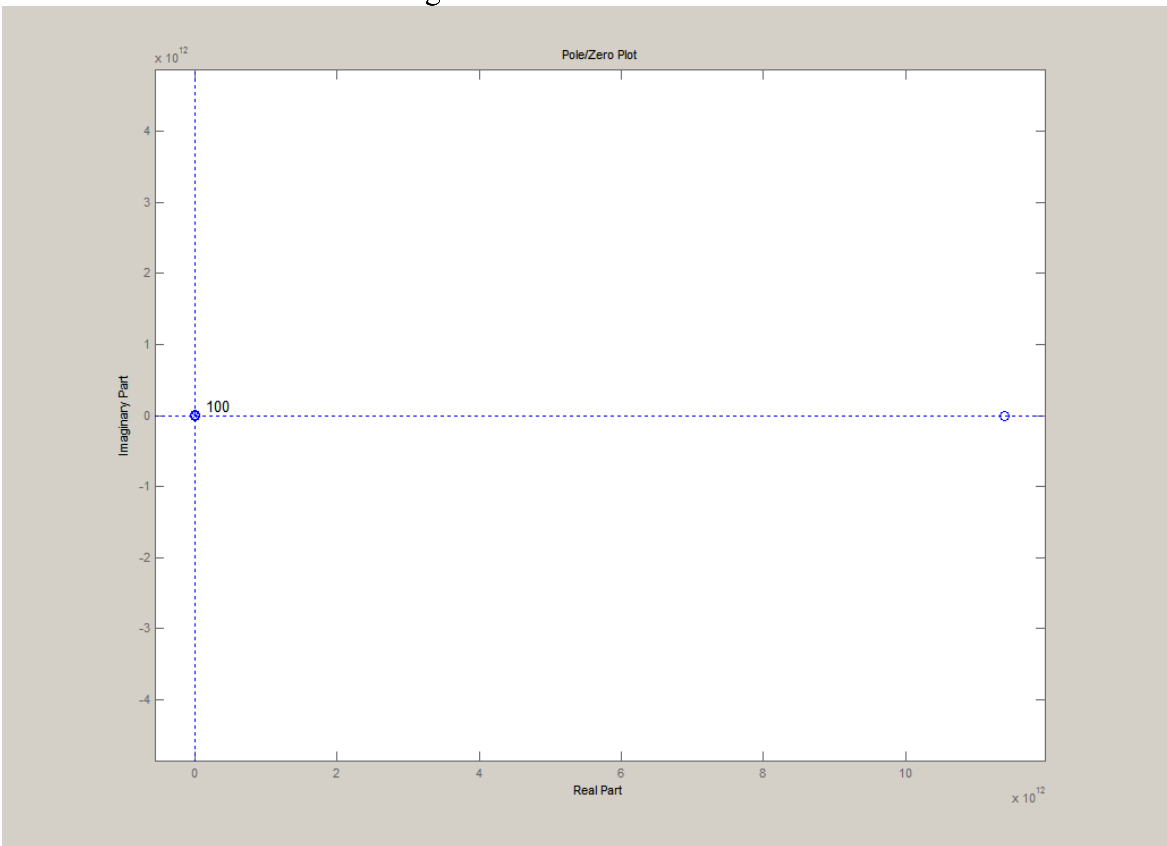


Fig.3.1g: MATLAB Simulation for 1-20 kHz FIR Blackman window Band-pass filter Magnitude and Phase Plot

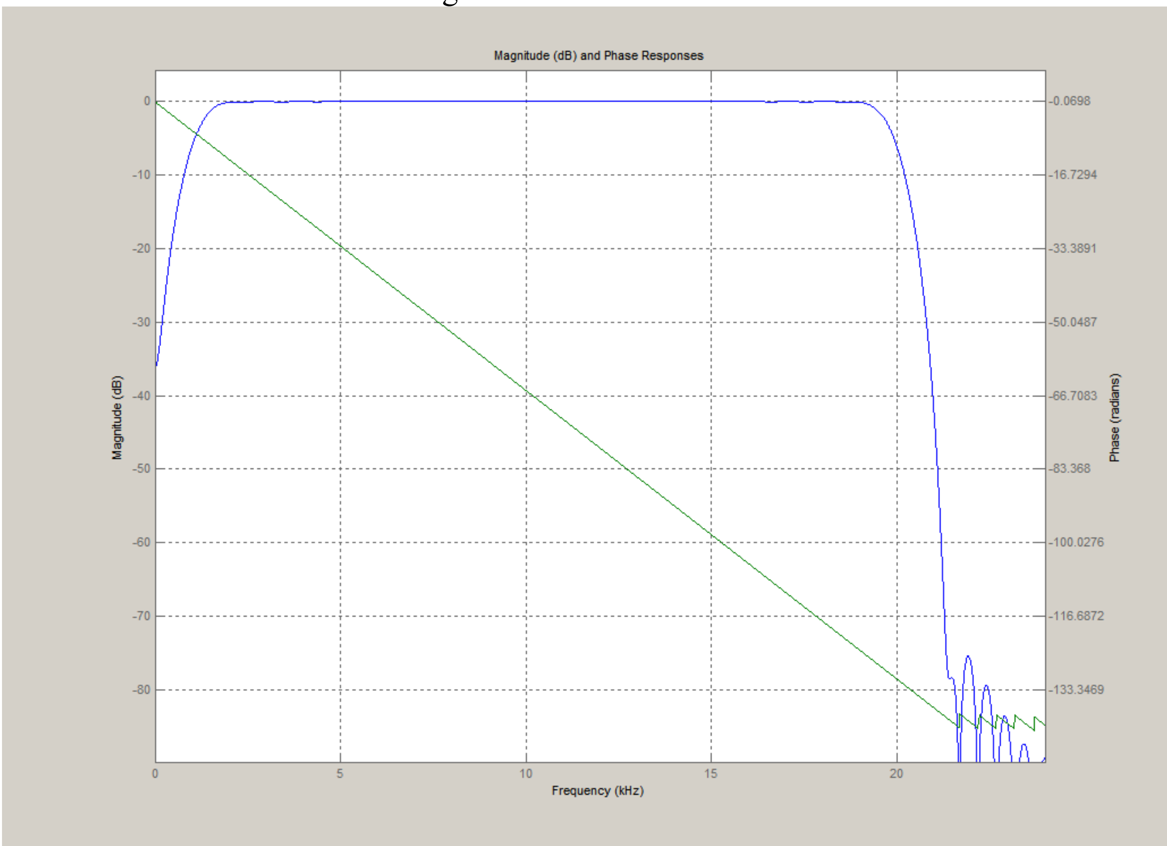


Fig.3.1h: Impulse Response Plot

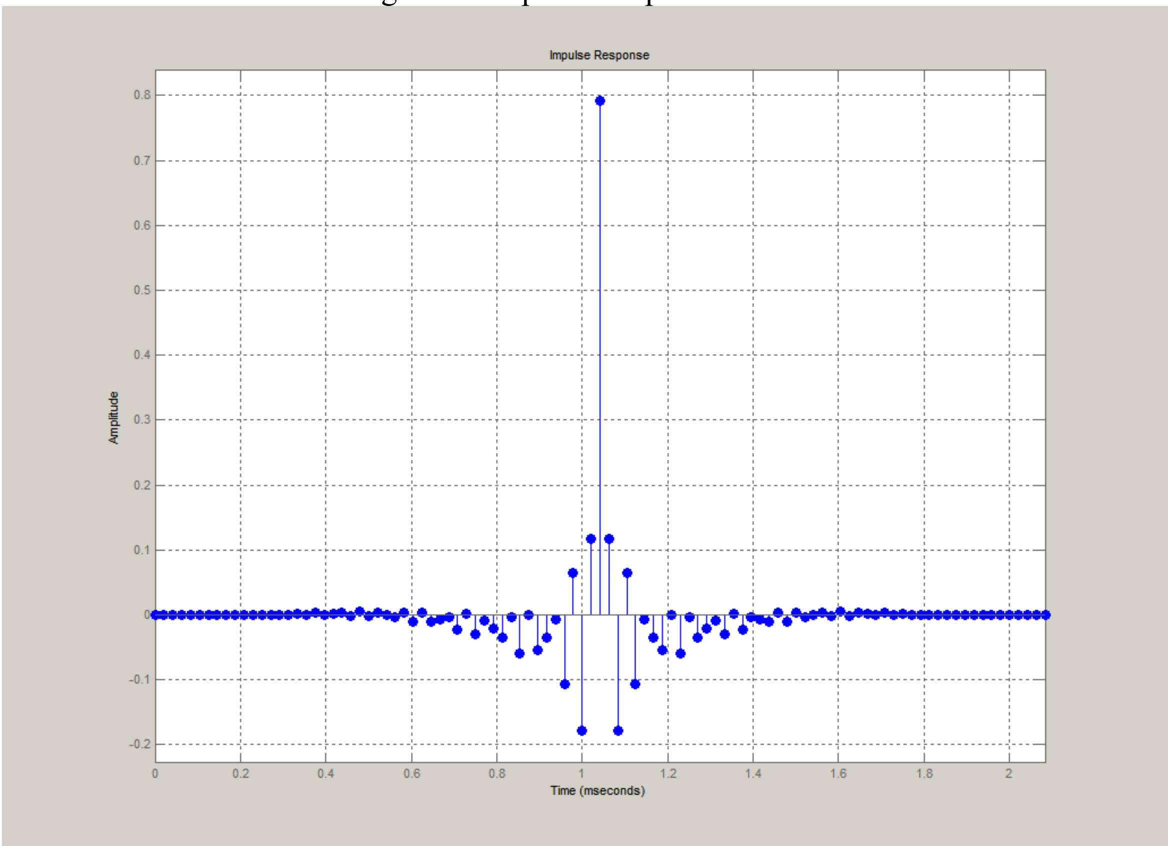
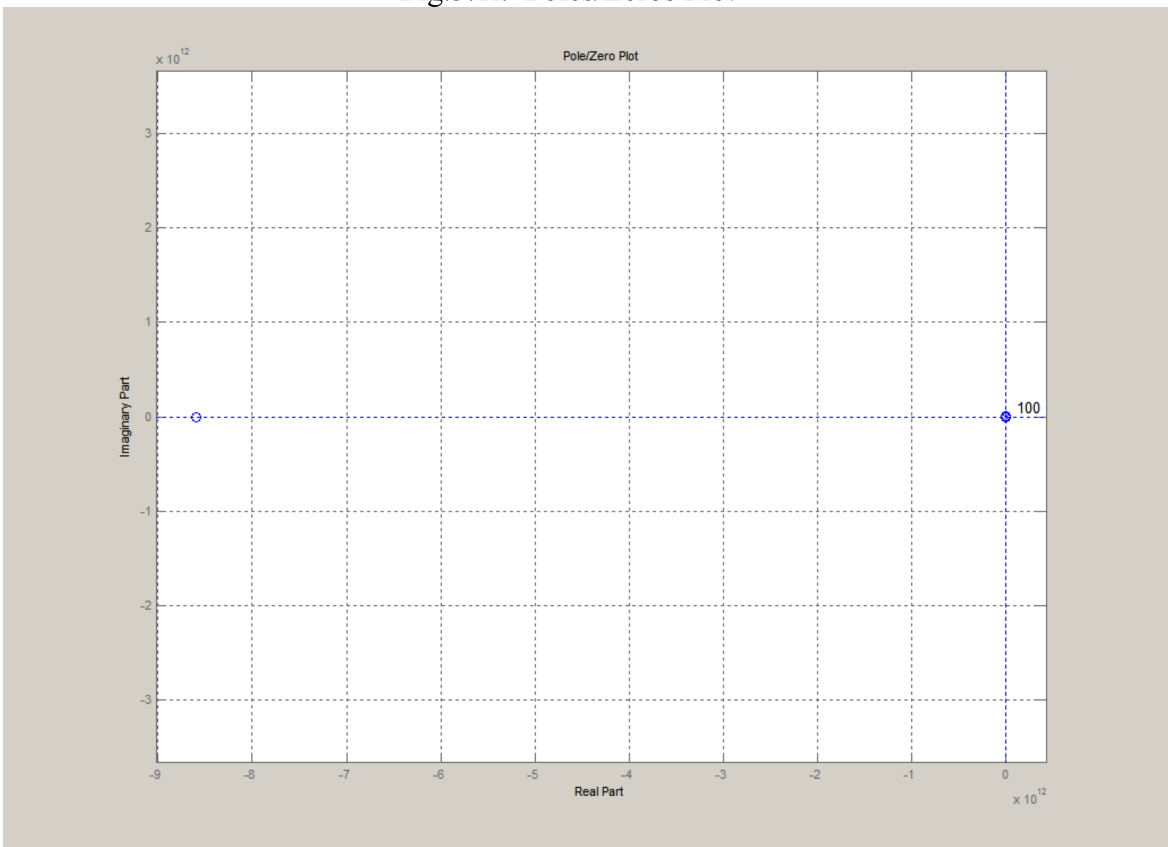


Fig.3.1i: Poles/Zeros Plot

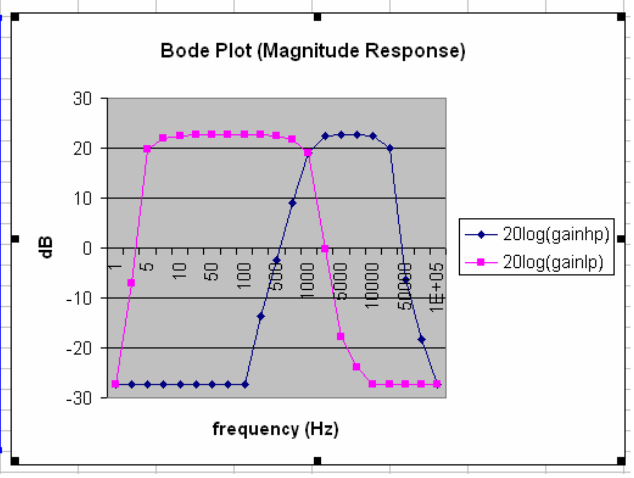


Calculated coefficients (Appendix 8) are exported into TI Code Composer Studio as header file. The looping code example was supplied with the Code Composer Studio (in C:\CCStudio_v3.1\examples\dsk5510\bsl\dsk_app).

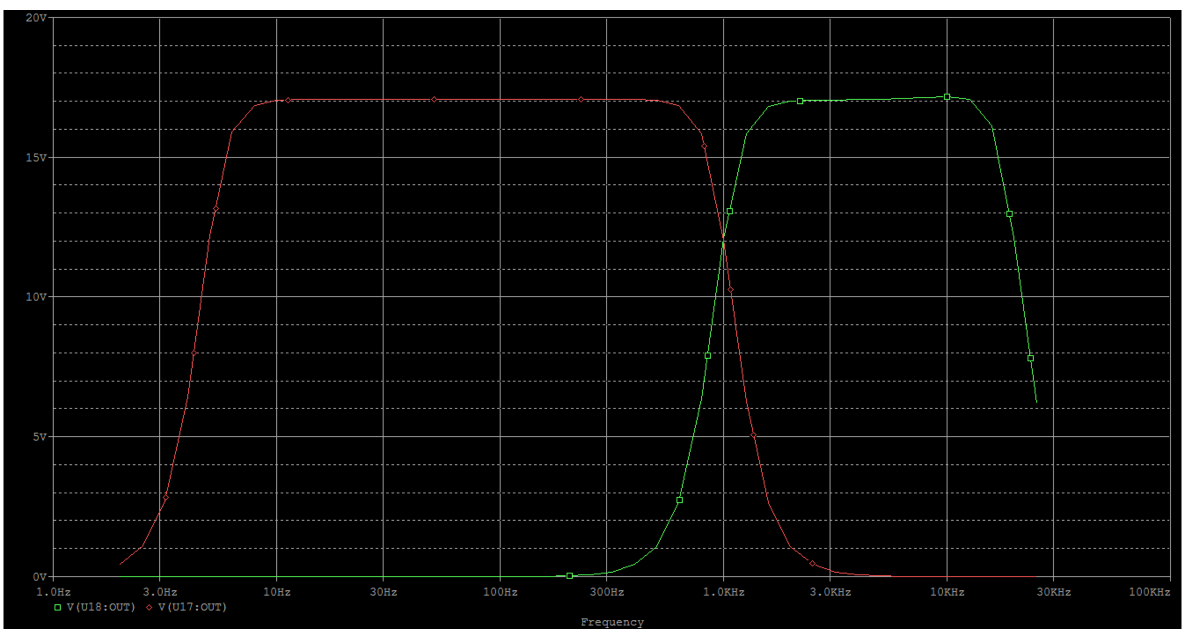
3.2 Experimental Results

After setting up our analog version of the filter, we connected the input and outputs of the circuit to an oscilloscope to measure its voltage gains. Using the recorded values, we use Microsoft Excel to graph its magnitude response.

freq (hz)	Vin-pp	Voutpp(hp)	Voutpp(lp)	20log(gainhp)	20log(gainlp)
1	0.233	0.01	0.01	-27.3471184	-27.3471184
2	0.233	0.01	0.102	-27.3471184	-7.17511499
5	0.233	0.01	2.27	-27.3471184	19.77339872
7	0.233	0.01	2.94	-27.3471184	22.01982819
10	0.233	0.01	3.11	-27.3471184	22.50808936
20	0.233	0.01	3.17	-27.3471184	22.67406682
50	0.233	0.01	3.17	-27.3471184	22.67406682
70	0.233	0.01	3.17	-27.3471184	22.67406682
100	0.233	0.01	3.17	-27.3471184	22.67406682
200	0.233	0.049	3.17	-13.5431968	22.67406682
500	0.233	0.177	3.06	-2.38765309	22.36731011
700	0.233	0.656	2.86	8.990958367	21.78020224
1000	0.233	2.09	2.09	19.0558073	19.0558073
2000	0.233	3.11	0.225	22.50808936	-0.30346806
5000	0.233	3.14	0.03	22.59147454	-17.8046933
7000	0.233	3.14	0.015	22.59147454	-23.8252932
10000	0.233	3.11	0.01	22.50808936	-27.3471184
20000	0.233	2.33	0.01	20	-27.3471184
50000	0.233	0.111	0.01	-6.44065884	-27.3471184
70000	0.233	0.0281	0.01	-18.372992	-27.3471184
100000	0.233	0.01	0.01	-27.3471184	-27.3471184



Below is the PSpice Simulation Magnitude Plot on page 14 (shown again below).



Comparing the bode plot magnitude responses of the experimental and the theoretical results, we see that we were able to achieve the expected values and curves of the ultra-low frequency filter. For the experimental magnitude response, we were able to recover a very similar shape in the curves of the low and high pass portions of the crossover filter compared to the theoretical one. The small variations in experimental results are a result of the limited number of frequencies we measured the voltage gains for and also the small difference in the actual resistor values used in the circuit (compared to the desired ones).

3.3 Demonstration Results

According to our experimental data, the actual audio signal should be separated by the designed analog filter without any problems. However, there was initially a lot of output noise, and static on the low-pass frequencies was heard when the speaker volume was raised too high. However, we later discovered that this was attributed to the poor quality speakers we bought. The generic speakers were not good enough to reproduce the low frequency sounds that a subwoofer normally specializes in. Later tests with higher quality speakers revealed that our crossover filter was working exactly as it was designed to, without too much interference noise.

The digital filter, on the other hand, was able to produce significantly better sound quality, even with the low quality speakers. Noise affects digital filters far less than analog ones, and thus the sound quality of the digital filter came out cleaner.

4. AREAS OF IMPROVEMENT

The final presentation of the crossover filter met design specifications and accomplished its task of splitting an audio signal at the desired crossover frequency. Although group members were quite content with the successful functioning of the designed circuit, there remains much room for improvement. A marketable product might have been achieved if manufacturability, economic issues, environmental issues, and health and safety concerns had also been considered.

4.1 Manufacturability and Economic Issues

The final working product appears as if it is still in the design process. A jumble of large resistors, capacitors, and wires stick out of two bulky breadboards. Although the work can be easily reproduced, with the aid of design layouts depicting each connection made onto the breadboard, the product is hardly manufacturable.

First off, more optimal parts could have been chosen for the crossover filter. Because the filter is not providing an amplified signal (external amplifiers for the speakers are required), current and voltage stresses should not be too high. In the design of the filter, we chose capacitors with voltage ratings of 500V, as well large resistors with a 1 Watt power rating. For the purpose of obtaining a working circuit that would not have any parts susceptible to burning out, the chosen voltage and power ratings were fine. On the economic standpoint, however, it would have been more efficient to obtain a range of maximum current and voltage stresses across the resistors and capacitors in the circuit, and than choose smaller parts that could still cope with these stresses.

Had a working filter been obtained earlier, the design could have been sent to a printed circuit board (PCB) making service, such as PCBExpress, to obtain a compact layout. The ability to mass produce PCBs and simply solder components onto the boards, without worrying about wires and loose connections at all, would have drastically improved the manufacturability of our crossover filter.

4.2 Environmental Sustainability and Safety Concerns

Our packaging container for the analog filter was just a simple cardboard shoebox. Such a presentation, although simple to produce, is unprofessional, flimsy, and not entirely safe. Should there be any spark accidentally generated by the circuit, the thin and flammable cardboard could easily house a fire and destroy the very thing it is supposed to protect.

The batteries and breadboard encased within the box did not have firm foundations, and so easily moved around. Combined with the loose connections of the breadboard, sudden movements or rough treatment of the packaging container would easily cause a wire to come loose and the circuit to function incorrectly.

Several things should have been performed to improve the stability and safety of the product. The components could have been soldered onto a perfboard or stripboard, instead of simply being stuck into a breadboard. Ideally, the components should have been soldered onto a compact printed circuit board. Instead of using a large, flimsy shoebox, we should have opted for a small, more form fitting plastic or wood container with foam cushioning on the inside, which would provide better protection against rough treatment of the product. Building an external battery compartment would also increase the marketability of the product.

5. TEAMWORK

During the two quarters that the project members had to work with one another, there were several class conflicts and inconveniences in meeting times and schedules. Initially, the group was not able to get together and meet with the faculty mentor until the 5^{th} week of first quarter.

Once the project members started to get together, however, they were able to make considerable progress in researching the topic of crossover filters. The book “Design of Analog Filters,” recommended by one of the project mentors, Fares, was taken from an interlibrary loan. Every page of the book was scanned, and a copy of the most instructive and informative chapters was distributed to each member of the group, so that all members would have a strong base knowledge in designing analog filters.

All members were inexperienced with PSpice, and cooperated in learning how to build and simulate each of the filter designs.

Each member of the group also brought distinguished traits that allowed the group to complete the project successfully.

Daniel’s ability to work in an organized, methodical manner was important in transposing the schematics onto the breadboard. The parts were laid down in a fashion that was simple and not confusing, which later on made debugging the circuit an easier process.

Kenneth had previous experience in designing circuits in an internship. He had a good idea of how to go about debugging a circuit when it did not work the way it was supposed to, and thus was instrumental in leading the debugging phase and achieving a working design.

Norihito was committed enough to the field of engineering to go out and purchase the most expensive equipment. He bought the Digital Signal Processor, and had the patience and diligence to successfully learn how to program it from scratch. Norihito spent the most time in learning and teaching the others about PSpice and analog filters.

6. COSTS

Table 7.1: Cost Analysis

Part	Quantity	Unit Price	***Subtotal***
TI TMS320VC5510 Fixed Point DSP (with built in A/D converters)	1	$395.00	$395.00
EXP-300 Socket/Breadboard (from Radio Shack)	1	$14.99	$14.99
EXP-300 Socket/Breadboard (from MarVac)	1	$8.75	$8.75
10nF Capacitor (25V Rating)	16	$0.166	$2.66
12.3KΩ Resistor (1/2W Rating)	4	$0.25	$1.00
16KΩ Resistor (1/2W Rating)	8	$0.19	$1.52
1.5KΩ Resistor (1W Rating)	4	$0.285	$1.14
3.3MΩ Resistor (1W Rating)	4	$0.65	$2.60
10kΩ Resistor (1W Rating)	8	$0.285	$2.28
800Ω Resistor (1/2W Rating)	4	$0.25	$1.00
3.5mm Cable Mini-plugs (Audio Jacks)	3	$2.295	$6.88
9V Batteries	4	$3.395	$13.58
Op Amp National LF351	8	$3.04	$24.32
Pocket Speaker/Amp	2	$12.99	$25.98
9V Battery Clips	2	$0.518	$1.04
Total			**$502.74**

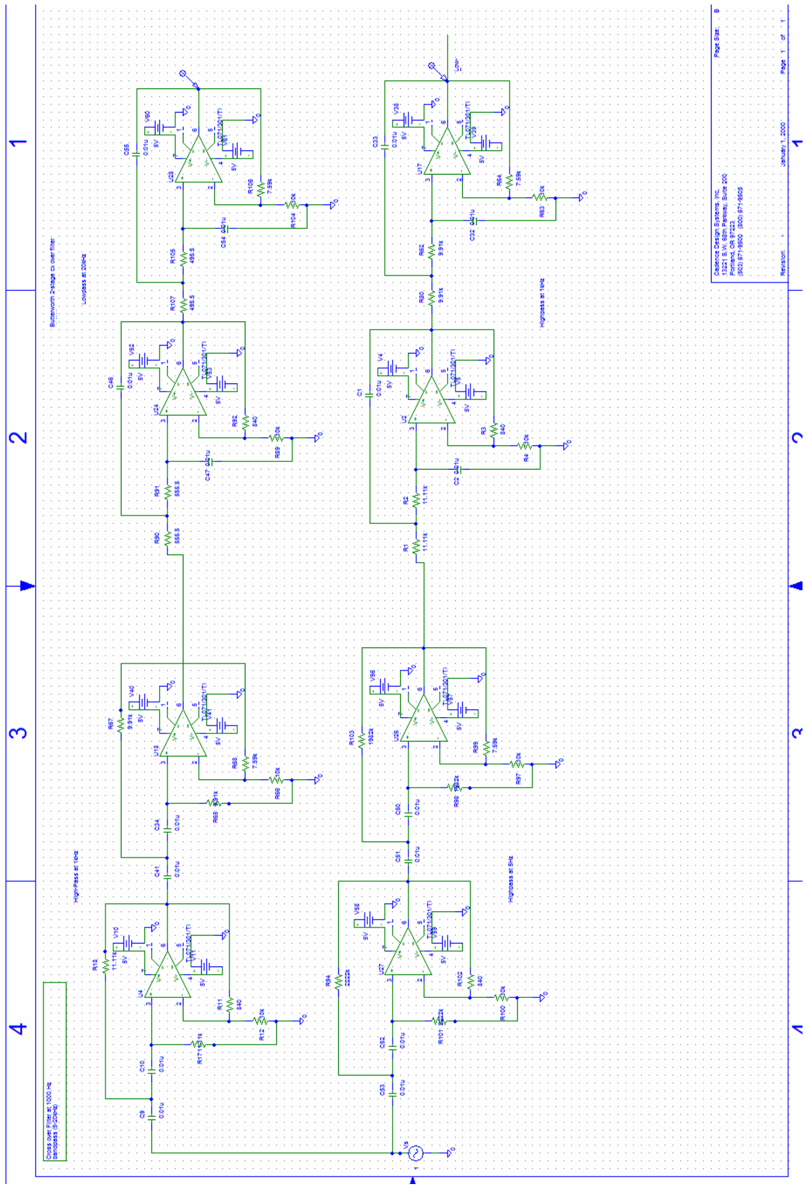
Other than the purchase of cheap audio speakers and the Digital Signal Processor, we were able to keep our budget well under $100.

7. SUMMARY

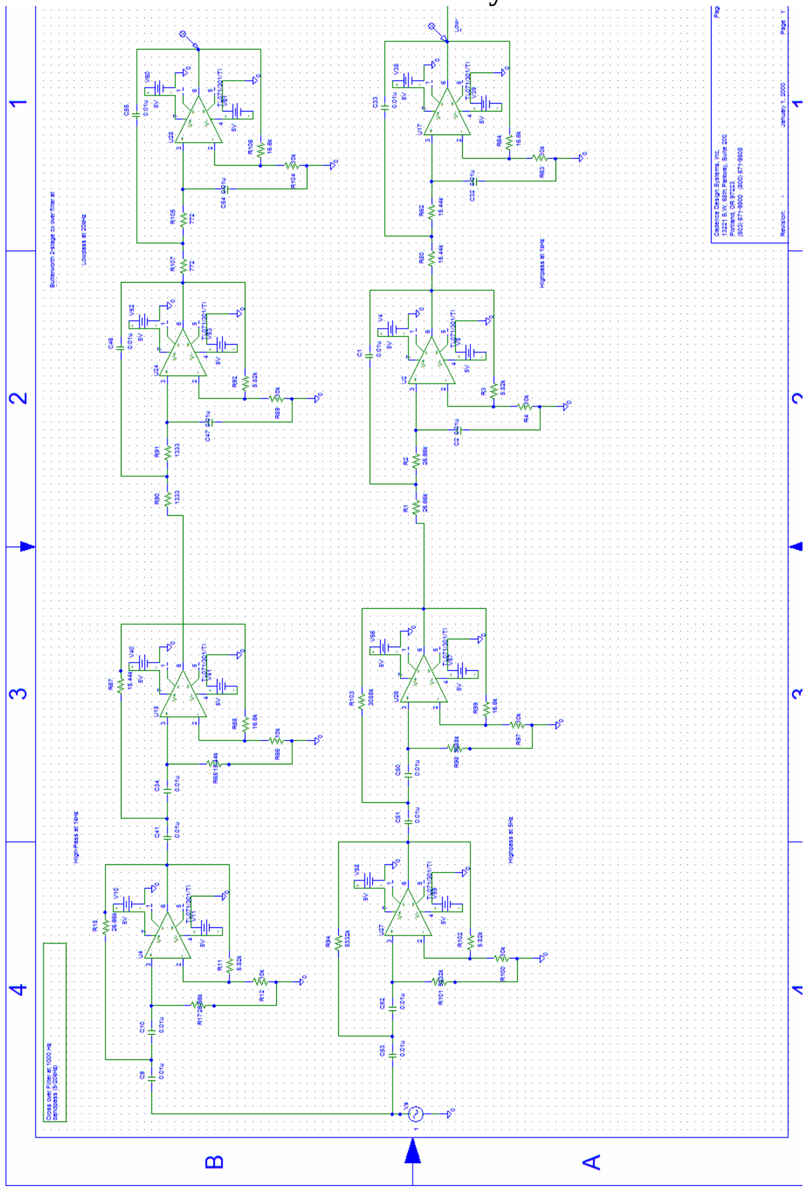
Although we did not build a very marketable product, our group was able to successfully learn about, design, and build a crossover filter from scratch. Group members learned how to use PSpice and how to design analog filters. More importantly, we became familiar with the design process of circuits: designing a satisfactory circuit, simulating the circuit to check if specifications are met, procuring the correct components, putting the circuit together, and debugging the circuit. Learning how to approach each of these steps in as efficient a manner as possible is fundamental in design engineering.

8. APPENDENCIES

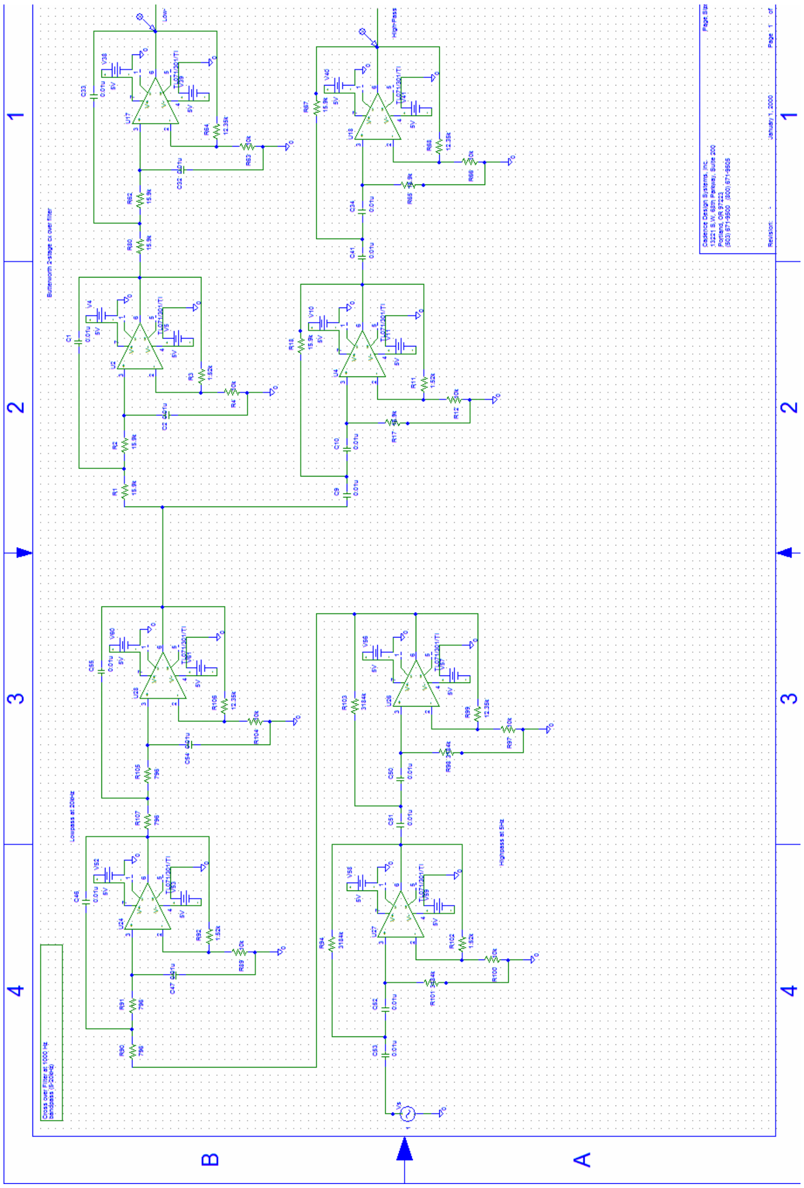
APPENDIX 1 – Schematic of Bessel Crossover Filter



APPENDIX 2 – Schematic of Chebyshev Crossover Filter



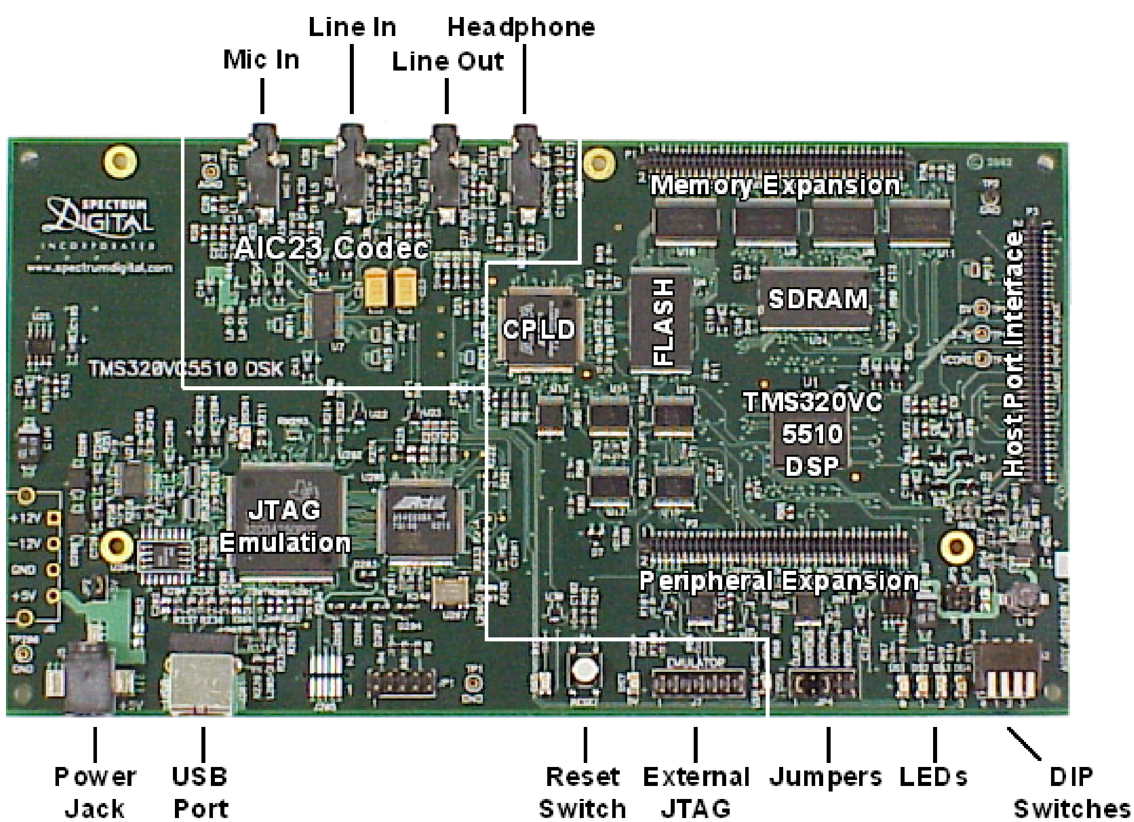
APPENDIX 3 – Schematic of Butterworth Crossover Filter



APPENDIX 4 – Texas Instruments TMS320VC5510 DSP Kit

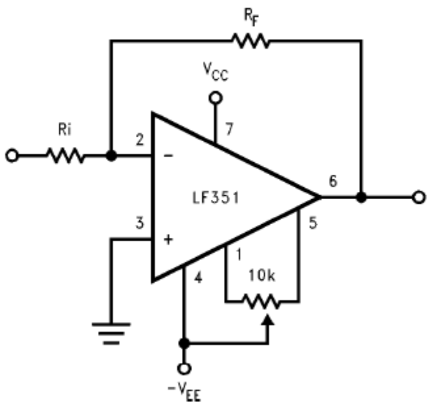
Hardware Features

- Texas Instrument's TMS320VC5510 DSP operating at 200 MHz.
- Embedded USB JTAG controller with plug and play drivers, USB cable included
- TLV320AIC23 codec
- 8 Megabytes of on board SDRAM
- 512K bytes of on board Flash ROM
- 3 Expansion connectors (Memory Interface, Peripheral Interface, and Host Port Interface)
- On board IEEE 1149.1 JTAG connection for optional emulator debug
- Four 3.5 mm. audio jacks (microphone, line-in, speaker, line-out)
- 4 user definable LEDs
- 4 position disp switch, user definable
- +5 Volt operation only, power supply included
- Size: 8.25" x 4.5" (210 x 115 mm), 0.062" thick, 6 layers
- Compatible with Spectrum Digital's DSK Wire Wrap Prototype Card



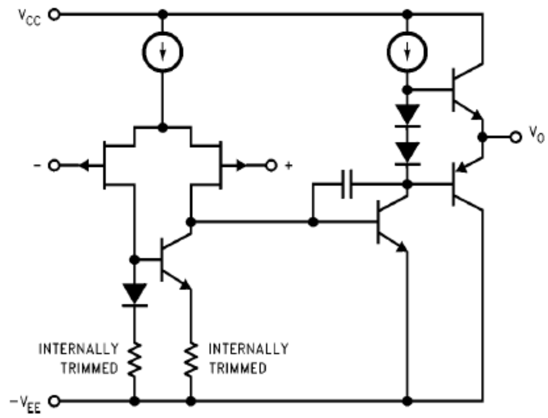
APPENDIX 5 – LF351 - Wide Bandwidth JFET Input Operational Amplifier

Typical Connection



TL/H/5648-11

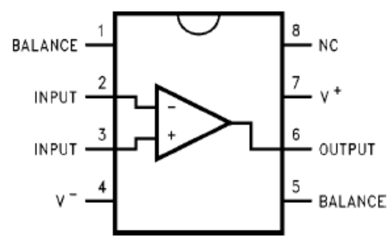
Simplified Schematic



TL/H/5648-12

Connection Diagrams

Dual-In-Line Package



TL/H/5648-13

Order Number LF351M or LF351N
See NS Package Number M08A or N08E

Features

■ Internally trimmed offset voltage	10 mV
■ Low input bias current	50 pA
■ Low input noise voltage	25 $nV/\sqrt{Hz}$
■ Low input noise current	0.01 $pA/\sqrt{Hz}$
■ Wide gain bandwidth	4 MHz
■ High slew rate	13 V/μs
■ Low supply current	1.8 mA
■ High input impedance	$10^{12}\Omega$
■ Low total harmonic distortion $A_V = 10$, $R_L = 10k$, $V_O = 20$ Vp-p, BW = 20 Hz–20 kHz	$<0.02\%$
■ Low 1/f noise corner	50 Hz
■ Fast settling time to 0.01%	2 μs

Absolute Maximum Ratings

If Military/Aerospace specified devices are required, please contact the National Semiconductor Sales Office/Distributors for availability and specifications.

Supply Voltage	$\pm$18V
Power Dissipation (Notes 1 and 6)	670 mW
Operating Temperature Range	0°C to +70°C
$T_{j(MAX)}$	115°C
Differential Input Voltage	$\pm$30V
Input Voltage Range (Note 2)	$\pm$15V
Output Short Circuit Duration	Continuous
Storage Temperature Range	−65°C to +150°C
Lead Temp. (Soldering, 10 sec.)	
Metal Can	300°C
DIP	260°C
θ_{jA}	
N Package	120°C/W
M Package	TBD
Soldering Information	
Dual-In-Line Package	
Soldering (10 sec.)	260°C
Small Outline Package	
Vapor Phase (60 sec.)	215°C
Infrared (15 sec.)	220°C

See AN-450 "Surface Mounting Methods and Their Effect on Product Reliability" for other methods of soldering surface mount devices.

ESD rating to be determined.

APPENDIX 6 – Software

The Mathworks MATLAB R2006a
http://www.mathworks.com/

Orcad PSpice Release 9.2
Orcad Layout Release 9.2
Orcad CIS Release 9.2
Orcad Capture Release 9.2
http://www.cadence.com/

Texas Instruments Code Composer Studio 3.1 IDE Platinum
http://focus.ti.com/dsp/docs/dspsupportatn.tsp?sectionId=3&tabId=415&familyId=44&toolTypeId=30

APPENDIX 7 – Polynomials

Butterworth, Chebyshev, and Bessel Polynomials. [6]

Table 1: Butterworth Polynomials

n	Transfer Function $H(s)$	Factored Form
1	$\frac{K}{s+\omega_c}$	
2	$\frac{K}{s^2+\sqrt{2}\omega_c s+\omega_c^2}$	
3	$\frac{K}{s^3+2\omega_c s^2+2\omega_c^2 s+\omega_c^3}$	$\left(\frac{K_1}{s+\omega_c}\right)\left(\frac{K_2}{s^2+\omega_c s+\omega_c^2}\right)$
4	$\frac{K}{s^4+2.61313\omega_c s^3+3.14142\omega_c^2 s^2+2.61313\omega_c^3 s+\omega_c^4}$	$\left(\frac{K_1}{s^2+1.848\omega_c s+\omega_c^2}\right)\left(\frac{K_2}{s^2+0.765\omega_c s+\omega_c^2}\right)$

Table 2: Chebyshev Polynomials

n	Transfer Function $H(s)$
1	$\frac{K}{s+1.9652\omega_c}$
2	$\frac{K}{s^2+1.0977\omega_c s+1.1025\omega_c^2}$
3	$\frac{K}{s^3+0.9883\omega_c s^2+1.2384\omega_c^2 s+0.4913\omega_c^3}$
4	$\frac{K}{s^4+0.9528\omega_c s^3+1.4539\omega_c^2 s^2+0.7426\omega_c^3 s+0.2756\omega_c^4}$

Table 3: Bessel Polynomials

n	Transfer Function $H(s)$
1	$\frac{K}{s+\omega_c}$
2	$\frac{K}{s^2+3\omega_c s+3\omega_c^2}$
3	$\frac{K}{s^3+6\omega_c s^2+15\omega_c^2 s+15\omega_c^3}$
4	$\frac{K}{s^4+10\omega_c s^3+45\omega_c^2 s^2+105\omega_c^3 s+105\omega_c^4}$

APPENDIX 8 – MATLAB: Results of the Filter Coefficients

1 kHz – 20 kHz Band-pass FIR Black Window Filter Coefficient

Direct-Form FIR Blackman Window Design Method
Order = 100, Stable, Sample frequency = 48kHz
The attenuation at cutoff frequencies is fixed at 6dB.

0,	0,	0,	0,	0,	-1,	-2,	-3,	-5,
-7,	-9,	-13,	-16,	-21,	-26,	-31,	-36,	-41,
-45,	-49,	-51,	-51,	-49,	-43,	-34,	-21,	-3,
21,	51,	87,	129,	179,	234,	297,	365,	438,
516,	597,	681,	766,	850,	932,	1011,	1085,	1153,
1213,	1264,	1305,	1334,	1352,	1359,	1352,	1334,	1305,
1264,	1213,	1153,	1085,	1011,	932,	850,	766,	681,
597,	516,	438,	365,	297,	234,	179,	129,	87,
51,	21,	-3,	-21,	-34,	-43,	-49,	-51,	-51,
-49,	-45,	-41,	-36,	-31,	-26,	-21,	-16,	-13,
-9,	-7,	-5,	-3,	-2,	-1,	0,	0,	0,
0,	0							

5 Hz – 1 kHz Band-pass FIR Black Window Filter Coefficient

Direct-Form FIR Blackman Window Design Method
Order = 100, Stable, Sample frequency = 48kHz
The attenuation at cutoff frequencies is fixed at 6dB.

0,	0,	0,	0,	1,	-1,	4,	0,	4,
10,	0,	26,	2,	31,	25,	15,	66,	-3,
88,	18,	49,	90,	-34,	149,	-75,	89,	0,
-116,	126,	-326,	97,	-332,	-238,	-111,	-738,	36,
-988,	-299,	-686,	-1160,	-88,	-1954,	0,	-1780,	-1159,
-218,	-3471,	2115,	-5828,	3848,	25941,	3848,	-5828,	2115,
-3471,	-218,	-1159,	-1780,	0,	-1954,	-88,	-1160,	-686,
-299,	-988,	36,	-738,	-111,	-238,	-332,	97,	-326,
126,	-116,	0,	89,	-75,	149,	-34,	90,	49,
18,	88,	-3,	66,	15,	25,	31,	2,	26,
0,	10,	4,	0,	4,	-1,	1,	0,	0,
							0,	0

9. REFERENCES

FROM BOOK

[1] Rolf Schaumann, and Mac E. Van Valkenburg, “Design of Analog Filters,” Oxford University Press, USA; 2Rev Ed edition, January 15, 2001

FROM THE INTERNET

[2] Steven W. Smith, “The Scientist and Engineer's Guide to Digital Signal Processing.” March 2007, http://www.dspguide.com/ch3.htm

[3] Wikipedia, “Butterworth filter.” March 2007, http://en.wikipedia,org/wiki/Butterworth_filter

[4] Wikipedia, “Chebyshev filter.” March 2007, http://en.wikipedia.org/wiki/Chebyshev_filter

[5] Wikipedia, “Bessel filter.” March 2007, http://en.wikipedia.org/wiki/Bessel_filter

[6] Salman Durrani, “ENGN3227 Analogue Electronics Problem Sets V1.0.” March 2007, http://engnet.anu.edu.au/DEpeople/Salman.Durrani/_teaching/ENGN3227_ProbSets.pdf