

Custom Design of a High-Fidelity Hearing Aid

1. Project Description

In this project you are asked to design a custom hearing aid filter that amplifies sound non-uniformly over the hearing spectrum to compensate for frequency dependent hearing deficiencies. One of your group members will be selected as the client for whom the hearing aid will be designed. Sound files and instructions for performing the hearing test to determine deviations from the *normal* threshold of hearing can be found at:

<http://www.engr.uky.edu/~donohue/audio/fsear.html>

Hearing loss typically occurs non-uniformly over the audible frequency range and varies from person to person. Clients must be tested with a series of tones that span the critical hearing spectrum to determine their unique hearing deficiencies. To evaluate a client's hearing response, incremental decreases in volume of a single tone (frequency) are made until the client no longer hears them. This test is often run in both directions, i.e. tones are also made incrementally louder until they can be heard. This is referred to as a test of limits, and running it in both directions reduces the bias or errors resulting from the subject's indication of when the sounds are audible/inaudible. The 2 results are then averaged to reduce biases in judgment. The threshold of hearing (TOH) level at each frequency is recorded as part of the hearing test. The hearing loss at the TOH can be quantified in terms of the gain required to bring the client's TOH to the normal threshold of hearing. This effectively defines the required transfer function (TF) magnitude for the custom filter circuit.

For example, at 70 Hz the normal TOH level is 30 dB, and a test shows the client has a TOH at 36 dB, which means the sound must be 6 dB louder than normal to be heard, and at 800 Hz the normal TOH level is 5 dB, but the client's TOH is 20 dB. Therefore, the filter and amplifier must provide a relative 6 dB gain at 70 Hz and a 15 dB gain at 800 Hz to restore the hearing sensitivity to that of a normal ear. After testing over the range of the hearing spectrum, specifications can be developed for the TF of an active filter circuit, which can be designed to compensate (as much as possible) for hearing losses at the tested frequencies.

Figure 1 shows a functional block diagram of the major components of the total hearing aid. The circuit to be designed for your project is just a part of the full hearing-aid device. The input to the total system is the sound impinging on the hearing-aid microphone, which converts changes in acoustic pressure to changes in voltage over the microphone terminal. The microphone output is fed into a broadband high-fidelity amplifier, which amplifies the weak signals from the microphone uniformly over the full audio range. The TF of the combined microphone and broadband amplifier are characterized by a flat spectral magnitude over the human hearing range. The next stage performs relative (not the absolute) amplifications and attenuations over the hearing spectrum to compensate for non-uniform hearing deviations from normal. The signal from this filtering circuit is sent to an audio amplifier to drive the earphone.

Assume the TF of the combined earphone and audio amplifier have a flat spectral magnitude for frequencies between 200 and 18000 Hz (includes the whole spectral range

of interest). The physical speaker element of the earphone cannot support a flat (high-fidelity) spectrum outside this range, so frequencies outside this range are not considered in the hearing test or filter design. A constant gain (volume) over this entire spectrum can be adjusted by the user through the audio amplifier driving the earphone. Thus, the absolute levels can be adjusted over the whole spectrum based on the user preference in their environment. The output impedance of the broadband amplifier has an output impedance of 50Ω and connects directly to the input of your circuit. The input impedance of the audio amplifier, which your circuit output is connected to, is $2\text{ M}\Omega$.

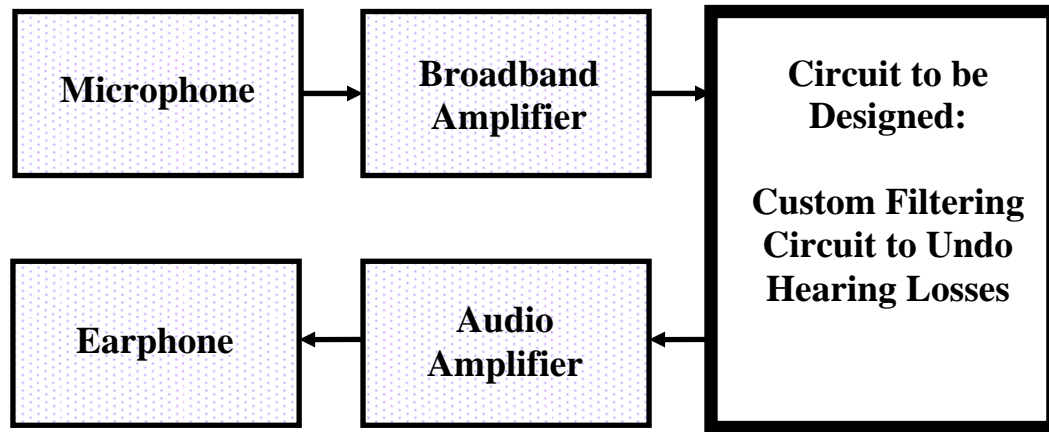
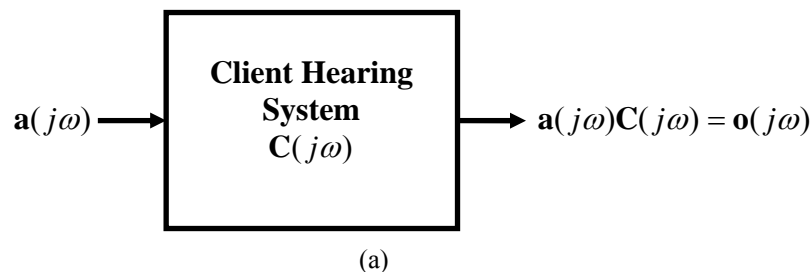


Figure 1. Top-level Functional Diagram of Hearing Aid.

2. Objective of the Hearing Aid Filter

The role of the hearing-aid filter is to effectively "correct" or scale the client's hearing sensitivity to achieve a normal TOH. Given an audio signal spectrum $\mathbf{a}(j\omega)$ and the client's hearing response (transfer function) $\mathbf{C}(j\omega)$, the hearing process can be expressed with block diagrams and equations as shown in Fig. 2a, where $\mathbf{o}(j\omega)$ is the psychophysical response (loudness) of the client to input $\mathbf{a}(j\omega)$. The ear can be thought of as a biological filter with input pressure wave (ear) and a perceptual output, which is a psychological response called *loudness*. This "hearing" process can be aided with an additional system, $\mathbf{H}(j\omega)$, cascaded between the pressure wave signal and the ear to undo deviations in the client response, $\mathbf{C}(j\omega)$, and achieve the desired or normal response, $\mathbf{n}(j\omega)$, as shown in Fig. 2b.



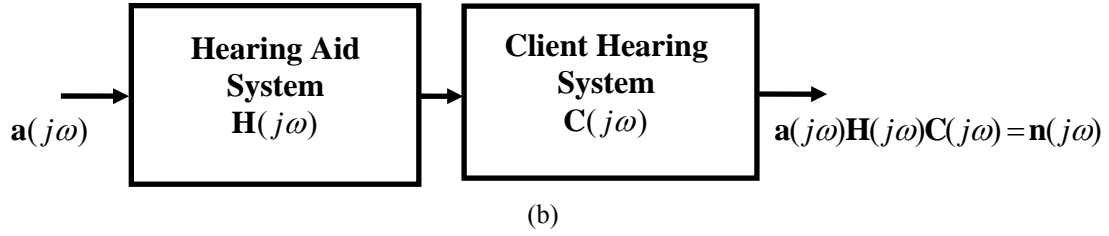


Figure 2. (a) System diagram of natural hearing process, input \mathbf{a} , output \mathbf{o} and transfer function \mathbf{C} . (b) System diagram of augmented hearing process.

The TF of the compensated system, shown in Fig. 2b, can be obtained by dividing both sides by the input to obtain:

$$\mathbf{H}(j\omega)\mathbf{C}(j\omega) = \frac{\mathbf{n}(j\omega)}{\mathbf{a}(j\omega)} = \mathbf{N}(j\omega) \quad (1)$$

where \mathbf{H} is the TF of the hearing aid filter circuit, \mathbf{C} is the TF of the clients ear, and \mathbf{N} is the TF of the ideal or normal psychophysical response. To obtain the TF required for the hearing-aid filter, Eq. (1) can be manipulated to obtain:

$$\mathbf{H}(j\omega) = \frac{\mathbf{N}(j\omega)}{\mathbf{C}(j\omega)} \quad (2)$$

Therefore, if the normal hearing TF is known (\mathbf{N}) and the client's hearing response measured (\mathbf{C}), Eq. (2) can be used to develop specification for the filter design. For this application the phase is not critical, so the design specifications only constrain the magnitudes of the TF. Thus, Eq. (2) can be expressed in terms of the TF magnitudes given by:

$$|\mathbf{H}(j\omega)| = \frac{|\mathbf{N}(j\omega)|}{|\mathbf{C}(j\omega)|} \rightarrow 20\log|\mathbf{H}(j\omega)| = 20\log|\mathbf{N}(j\omega)| - 20\log|\mathbf{C}(j\omega)| \quad (3)$$

Equation (3) indicates that only differences between the normal and client magnitude response are needed to develop the hearing aid circuit specification. This suggests a process that is simpler than actually getting the TF of the ear, which involves only measuring the client TOH at a discrete number of frequencies.

The TOH is defined as the smallest audio power level at which a person can *still hear* a signal, based on an absolute scale. These levels over the audio spectrum form a function that corresponds to the reciprocal of the TF. Since calibrated equipment will not be used on the test, only relative level will be determine. To examine the relationship between absolute and relative TOH functions, let $\bar{\mathbf{N}}$ be the absolute normal TOH, and $\bar{\mathbf{C}}$ be the absolute TOH of the client. The transfer function from a relative TOH measurement is described by the following equations:

$$\mathbf{N} = \frac{\alpha}{\bar{\mathbf{N}}} \quad \mathbf{C} = \frac{\beta}{\bar{\mathbf{C}}} \quad (4)$$

where α and β are proportionality constants that have no impact on the shape (relative changes) of the TF. These result from different volume level settings on the playback devices and listening station. While the true TF values require absolute measurement of sound, the absolute scale is not needed in our case since the device will have a volume adjustment for the client to change overall level of the signal. So the important aspect of our design is the relative differences or shape of the circuit TF. This is observed by substituting the TOH function from Eq. (4) into Eq. (3) to obtain:

$$|\mathbf{H}(j\omega)| = \frac{|\bar{\mathbf{C}}(j\omega)\alpha|}{|\bar{\mathbf{N}}(j\omega)\beta|} \rightarrow 20\log|\mathbf{H}(j\omega)| = 20\log|\bar{\mathbf{C}}(j\omega)| - 20\log|\bar{\mathbf{N}}(j\omega)| + 20\log\left|\frac{\alpha}{\beta}\right| \quad (5)$$

where the last term is a constant resulting from the proportionality factors and would be directly affected by the overall volume control and can be ignored for our circuit. So Eq. (5) indicates that the measured TOH level of the client's ear in dB, minus the normal TOH level is our circuit specification.

For this project the client's threshold of hearing will be measured at discrete frequencies contained in the Table 1.

Table 1. Test Frequency hearing thresholds

Test Frequencies (Hz)	Absolute TOH (dB)
300	9
600	6
1200	4
2400	0
4000	-4
5000	0
7000	10
9000	18
12000	11
14000	20

The set of frequencies are assumed to be dense enough in the critical part of the spectrum, such that they provide sufficient information for the continuous reconstruction of the filter over the critical frequency range. At each frequency the tone is played with decreasing amplitudes (3 dB decrements). The client will count the number of levels that can be heard. Many levels counted imply good sensitivity (ability to hear low-level sounds). Each count represents a relative 3 dB drop in acoustic level.

The highest tone count will typically occur around 4000 Hz, which is the most sensitive region in the hearing spectrum. The level at this point will be set to 0 by subtracting the count at 4000 Hz from all client test frequencies. This makes the counts and sensitivity levels at all other frequencies relative to the sensitivity at 4000 Hz. Since each subsequent tone is 3 dB less than the previous, scale the count values by 3 dB (i.e. multiply all the

count values by 3). Now subtract 4dB from the dB levels all the frequencies to result in the 4000 Hz level being set to -4dB (which is what it is approximately at the absolute normal TOH). After this scaling, these levels can now be compared directly to the normal TOH response. Results should match at 4000 Hz (since this was the reference point for the subtraction and scaling), and the values at other frequencies represent the relative amplification needed to match the shape of the client ear to the normal ear. The subtraction and scaling described above compensates for the proportionality constants in Eq. (5).

3. Performance Optimization and Additional Constraints

Engineering activities involve finding system parameters to meet design specifications (constraints) and optimizing performance. To help with this process (especially for complex systems) the constraints and performance metrics should be expressed quantitatively (mathematical models). This allows the design process to use efficient mathematical analyses and computer programs to search for parameters resulting in the best possible performance as defined by the model and design specification.

In this application, closeness between the TF magnitudes derived from the TOH measurements and the designed filter is directly related to performance. This closeness can be mathematically formulated to become an objective function to be minimized (minimize error to maximize performance). Two metrics will be useful for quantifying this difference, namely the bias error and rms error. The bias error represents the average difference between all the points in target TF (i.e. a level shift). The bias error in terms of the circuit TF is given by:

$$\mu = \frac{1}{N} \sum_{k=1}^N 20 \log |\mathbf{H}_C(j2\pi f_k)| - 20 \log |\mathbf{H}_D(j2\pi f_k)| \quad (6)$$

where \mathbf{H}_C is the measured TF for the circuit, \mathbf{H}_D is the desired TF derived from the client, f_k is a test frequency, and N is the number of test frequencies. Note that this is an average distance between the levels of the 2 TFs. So for example, if μ is computed to be 3.2 dB, the circuit TF has an overall gain of 3.2 dB greater than the desired TF. In this case an all-pass amplifier stage can be set to reduce the gain by 3.2 dB on the designed circuit. Then μ will be zero (or close to it). So if there is an amplifier stage in the design where this adjustment can be done for all frequencies, the μ value can almost always be set to zero with no additional circuit complexity.

The rms error represents the overall shape difference between the 2 TFs, and is more difficult to minimize. The rms error is given by:

$$\sigma = \sqrt{\frac{1}{N} \sum_{k=1}^N (20 \log |\mathbf{H}_C(j2\pi f_k)| - 20 \log |\mathbf{H}_D(j2\pi f_k)| - \mu)^2} \quad (7)$$

where μ is the bias error computed in Eq. (6). Unlike Eq. (6) the difference terms of this summation are always positive because the difference is squared with the overall level difference subtracted out to emphasize shape difference. It does not matter whether

deviations are positive or negative, they contribute directly to the error and DO NOT cancel each other out over the summation (as happens with the bias error computation).

Equations (6) and (7) represent objective functions to be minimized for maximum performance. There are 2 other filter performance constraints that must be considered for energy outside the test frequency range. The signals outside this range must be attenuated, so the out-of-band energy does not saturate or drain battery power by driving the output amp with frequencies that are not important to hearing. So the transfer function of the designed circuit must be rolling-off (getting smaller) for frequencies smaller than 100 Hz and greater than 18000 Hz. And the roll-off must be fast enough so that the transfer function at 100 and 18000 Hz must be less than 3 dB and decreasing (the smaller the better). The out-of-band level constraint can be expressed mathematically as:

$$20 \log |\mathbf{H}_D(j2\pi 100)| < 3 \quad \text{and} \quad 20 \log |\mathbf{H}_D(j2\pi 18000)| < 3 \quad (8)$$

The roll off requirement can be stated in terms of the slope being positive for frequencies approaching 0 and negative for frequencies approach infinity:

$$\begin{aligned} \frac{d(20 \log |\mathbf{H}_D(j2\pi f)|)}{df} > 0 \quad \text{for} \quad 0 < f < 100 \\ \frac{d(20 \log |\mathbf{H}_D(j2\pi f)|)}{df} < 0 \quad \text{for} \quad f > 18000 \end{aligned} \quad (9)$$

These specification are referred to as the *out-of-band constraints* since they apply to frequencies outside of the test range.

4. Limitations on Circuit Components

The constraints and objectives discussed in the previous section can be met through frequency selective filters (i.e., low pass, high pass, band pass, or band reject) combined in an appropriate manner. Both active and passive filters can be used. The following are limitations on the components used construct the filter:

- 1) *The design is limited to use at most 6 op amps* due to power consumption and chip space.
- 2) Values for capacitors, resistors, and inductors must be available from off the shelf parts (parts lists are available in Appendix A). Models for op amps in the final design should be the 5-pin parameterized op amp in B² Spice.

The process for your design project should follow these steps:

1. **Determine the ideal transfer function for your circuit** at the frequency test points and concisely state your design problem, including ALL additional critical constraints (out of band criteria, part limits ...). The document should be complete enough so that the engineer solving the problem does not need additional information regarding the circuit performance/specifications. Use tables, graphs, and equations as appropriate to maximize precision of the specifications and minimize the number of words.
2. **Develop a top-level description for your design** by combining basic filter responses in appropriate configurations to realize the circuits TF magnitude. The

- basic filters include the low-pass, high-pass, band-pass, and band-reject filters, which can be cascaded to effectively multiply their TF magnitudes and/or arranged in parallel (effectively adding the TFs) with a summing circuit. A graph of the desired TF magnitude is helpful here. From the shape of the graph you should be able to propose an initial combination of filters to approximate the desired shape.
3. **Simulate the proposed top-level circuit with mathematical models of the TFs in Matlab.** Matlab scripts and functions can be used to combine the TF models of the subcircuits to get the overall TF and compare it to the target at the test points. In this part the filter parameters are determined independent of the specific circuit components that will be used to implement the filter. Use Matlab's iteration and graphic/visualization capabilities to *optimize* the filter parameters such that the bias and rms errors are minimized, while the out-of-band constraints are met.
 4. **Find actual circuits to implement filters and simulate your circuit using SPICE** (i.e. find the circuit parameters that come closest to the derived ones using the off-the-shelf components of Appendix I).

After finalizing the circuit use results from SPICE, to verify that your design meets all proposed specifications and present the performance metrics (μ and σ) for the optimized design. Performance parameters from the final Matlab optimization should be presented as well for comparison (i.e. performance metrics of the optimized filter parameters in Matlab to those obtain from the simulation in SPICE).

Proposal Report:

1. Cover Page
2. (Section 1) Definition of the problem and a complete set of specifications associated with your client. A description of how the target TF was obtained (i.e. describe the experiment or test used to get the numbers reported). This description should include a graph of your client's threshold of hearing compared to the normal threshold of hearing, a graph of the TF for the circuit to be designed, and a table with the required gain values at each frequency. It should also include any other conditions required of your circuit.
3. (Section 2) A preliminary top-level design with a succinct description of each functional block (i.e. What does it accomplish relative to the general target TF shape? What are the approximate filter parameters for each functional block or what frequency range is the block primarily intended to influence?).
4. (Section 3) A time-table for completing the project with specific target dates identified for task completion, and assignment of responsibilities for each team member. Note that this requires thinking through all the tasks of the project - including writing the reports.

Note that you will *not* be bound by the top-level design presented in the proposal report. As you begin to design and test the filters, you may discover a better approach for designing your circuit. Your preliminary design will serve as your starting point and an opportunity for the instructor to look over your approach and give comments. However, you cannot change your client specifications after the preliminary report is submitted.

Final Report:

The final report will be a final summary of your final design. It should contain the following:

1. Cover Page
2. Include Sections 1-3 of the Phase I report, with corrections and updates as necessary.
3. (Section 3) Description of the optimization process.
 - a. Description of optimization process
 - b. Presentation of optimized design (include graph of designed TF with target TF)
 - c. Quantification of mean and rms errors
 - d. Validation that design meets out-of-band design specifications

State the design metrics used and the final outcome as measured by the mean and rms errors. Discuss difficulties and successes of the optimization process, and alternatives tried. Provide the commented codes in an appendix. This description should include a graph of circuit TF based on your optimized parameters compared to the target transfer function. You must also discuss (from the plot) how well your TF met the out-of-band constraints. I must see conclusive statements regarding your design and constraints in the specification.

4. (Section 4) Final Circuit Performance
 - a. Circuit design for each function block of the top-level design.
 - b. Schematic of the final circuit.
 - c. Compare graphically the final circuit TF against the target TF (to do this, you can get the values of the transfer function from SPICE by outputting a table of values).
 - d. Compute the mean and rms errors of the SPICE circuit model with that obtained in the Matlab optimization. Discuss reasons for differences.

The circuit parameters must be clearly shown as well as a comparison between the performance in the SPICE and Matlab simulations. I must see conclusive statements regarding your design and constraints in the specification for the SPICE simulation.

Prepare the reports on a word processor (MS Word, Open-Office ...) and email them to me before the start of class on the due date. The reports will be graded based both on technical presentation and grammar (sentence structure, organization, and spelling). The reports should be *succinct*. Avoid redundancy. Also, avoid superfluous information (better known as "fluff"). Remember that the audience that you are writing is *not* your instructor, it is an engineer who will need to read your report, understand your design, and build the circuit, or do further development. Thus, the reader does not need to know the history of hearing aids. Rather, they will need a brief - yet thorough - description of the filter design process. Finally, since the report is a technical report, it should be void of personal pronouns such as "I" or "we." Each group must email me *one* copy of the report representing all team members. All members of the group will hand in their individual engineering notebooks for the project.

Engineering Notebooks

Each member of the group is expected to keep an engineering notebook. This is a technical diary that can be used to track the evolution of the project. The notebooks required for this assignment are the 48-page student lab books, which can be purchased from Eureka <http://www.eurekalabbook.com/Studentnote.html> or from the IEEE parts store here on campus <http://www.engr.uky.edu/~ieee>.

Guidelines for maintaining the notebook:

- 1) Date and sign each new entry
- 2) All handwritten work done for the project should be done in the notebook. For example, all brainstorming, calculations, mathematical derivations or algebraic manipulations, etc. should be done in the notebook. If mistakes are made, cross them

out. Do not tear pages out of the notebook. It is equally important to keep records of mistakes as well as successes. Mistakes can often provide important information and authenticity of the thought process used to develop the device. All work in the notebook should be explained. For example, if you analyze a circuit, you should start out by stating what you are looking for and why. Short comments and introductory sentences are critical for others to follow the work contained in your notebook. You should assume other engineers will need to read this as well as patent lawyers. Each page should be dated and signed.

- 3) Circuit diagrams, data and graphs should be printed out and cut and pasted (or stapled or taped) in the notebook. Also, computer codes written in Matlab or using other software or languages should be printed out and cut and pasted into the notebook. In general a flow chart should be used to describe software/code that is being developed, in addition to clear statements of the purpose of the code and its inputs and outputs.
- 4) Records of group meetings should be kept in the notebooks. Record general discussions and decisions made at the meeting.
- 5) Try to be neat and orderly in your notebook, since you and others may need to reference back to this information at a later date.

The engineering notebooks of each team member will be handed in with the final report. There is no excuse for a late engineering notebook. You will lose points for a late engineering notebook. All work presented must be in chronological order (i.e. do NOT give a series of short statements of what you did each day, and then stuff all your derivations, programs, plots, etc. in the back of the notebook). Points will be lost if it APPEARS that the notebook was put together after the fact.

The following pages are the cover pages that will be handed in with your reports.
Reports can be submitted via email to the instructor. If for some reason you cannot incorporate the cover page with your report, you can email it as a separate document.

Proposal Report Cover Page

Group Member: _____ Date: _____

Group Member: _____ Date: _____

Group Member: _____ Date: _____

Group Member: _____ Date: _____

Section I - Problem Definition _____ (/1 points)

- A. Brief problem definition
- B. TOH for client and target TF for circuit design
- C. Complete design specifications/constraints

Section II - Top-Level Design _____ (/1 points)

- A. Presentation of block diagram of top-level design
- B. Description of the purpose and function of each block. Description should convince reader that design is compatible with target TF.

Section III - Time-Table _____ (/3 points)

- A. Break-down of project into smaller sub-tasks
- B. Distribution of effort (primary responsibility for each member)
- C. Tentative time schedule for sub-tasks in order to complete project by required date

Preliminary Total: _____ (/5 pts)

Format × _____ % (/100 %)

- A. Technical presentation
- B. Grammar, Spelling, Succinctness

Total: _____ (/5 pts)

Final Report Cover Page

Group Member: _____ Date: _____
Group Member: _____ Date: _____
Group Member: _____ Date: _____
Group Member: _____ Date: _____

Section I - Problem Definition (From Proposal report) _____ (/2 points)

- A. Brief problem definition
- B. TOH for client and target TF for circuit design
- C. Complete design specifications/constraints

Section II - Top-Level Design (From Proposal report) _____ (/2 points)

- A. Presentation of block diagram of top-level design
- B. Detailed description of the function of each block (include filter parameters i.e. bandwidth, resonant frequency, cut offs, gains ...). Description should convince reader that design is compatible with target TF.

Section III – Design of Transfer Function (From Proposal report) _____ (/3 points)

- A. Description of optimization process
- B. Presentation of optimized design (include graph of designed TF with target TF)
- C. Quantification of mean and rms errors
- D. Validation that design meets out-of-band design specifications

Section IV – SPICE Model _____ (/4 points)

- A. Circuit design for each function block
- B. Final SPICE design & quantification of mean and rms errors
- C. Comparison of SPICE design with target TF (overlay graph)
- C. Comparison of mean and rms errors (use table to present the numbers)
- E. Verification that design meets all specifications in problem statement

Total: _____ (/11 pts)

Format × _____ % (/ 100 %)

- A. Technical presentation
- B. Grammar, Spelling, Succinctness

Final Report Total: _____ (/11 pts)

Proposal Report Score: _____ (/5 pts)

Total Project Group Grade: _____ (/16 pts)

Engineering Notebook: Worth a total of 6 points.

Appendix A: Parts List

Surface Mount Capacitors

Capacitance	Tolerance	Price each	Capacitance	Tolerance	Price each
4.7uF	10%	0.37	4.7uF	5%	0.47
3.9uF	10%	0.37	3.9uF	5%	0.47
3.3uF	10%	0.37	3.3uF	5%	0.47
2.7uF	10%	0.37	2.7uF	5%	0.47
2.2uF	10%	0.37	2.2uF	5%	0.47
1.8uF	10%	0.37	1.8uF	5%	0.47
1.5uF	10%	0.37	1.5uF	5%	0.47
1.2uF	10%	0.37	1.2uF	5%	0.47
1.0uF	10%	0.37	1.0uF	5%	0.47
0.82uF	10%	0.35	0.82uF	5%	0.45
0.68uF	10%	0.35	0.68uF	5%	0.45
0.56uF	10%	0.35	0.56uF	5%	0.45
0.47uF	10%	0.35	0.47uF	5%	0.45
0.39uF	10%	0.35	0.39uF	5%	0.45
0.33uF	10%	0.35	0.33uF	5%	0.45
0.27uF	10%	0.35	0.27uF	5%	0.45
0.22uF	10%	0.35	0.22uF	5%	0.45
0.18uF	10%	0.35	0.18uF	5%	0.45
0.15uF	10%	0.35	0.15uF	5%	0.45
0.10uF	10%	0.35	0.10uF	5%	0.45
0.082uF	10%	0.35	0.082uF	5%	0.45
0.068uF	10%	0.35	0.068uF	5%	0.45
0.056uF	10%	0.35	0.056uF	5%	0.45
0.047uF	10%	0.35	0.047uF	5%	0.45
0.039uF	10%	0.35	0.039uF	5%	0.45
0.033uF	10%	0.35	0.033uF	5%	0.45
0.027uF	10%	0.35	0.027uF	5%	0.45
0.022uF	10%	0.35	0.022uF	5%	0.45
0.018uF	10%	0.35	0.018uF	5%	0.45
0.015uF	10%	0.35	0.015uF	5%	0.45
0.010uF	10%	0.35	0.010uF	5%	0.45
0.0082uF	10%	0.33	0.0082uF	5%	0.43
0.0068uF	10%	0.33	0.0068uF	5%	0.43
0.0056uF	10%	0.33	0.0056uF	5%	0.43
0.0047uF	10%	0.33	0.0047uF	5%	0.43
0.0039uF	10%	0.33	0.0039uF	5%	0.43
0.0033uF	10%	0.33	0.0033uF	5%	0.43
0.0027uF	10%	0.33	0.0027uF	5%	0.43
0.0022uF	10%	0.33	0.0022uF	5%	0.43
0.0018uF	10%	0.33	0.0018uF	5%	0.43
0.0015uF	10%	0.33	0.0015uF	5%	0.43
0.0010uF	10%	0.33	0.0010uF	5%	0.43
0.00082uF	10%	0.31	0.00082uF	5%	0.41
0.00068uF	10%	0.31	0.00068uF	5%	0.41
0.00056uF	10%	0.31	0.00056uF	5%	0.41
0.00047uF	10%	0.31	0.00047uF	5%	0.41
0.00039uF	10%	0.31	0.00039uF	5%	0.41
0.00033uF	10%	0.31	0.00033uF	5%	0.41
0.00027uF	10%	0.31	0.00027uF	5%	0.41
0.00022uF	10%	0.31	0.00022uF	5%	0.41

Surface Mount Resistors:**Base Factors:**

1.0, 1.1, 1.2, 1.3, 1.5, 1.6, 1.8, 2.0, 2.2, 2.4, 2.7, 3.0, 3.3, 3.6, 3.9, 4.3, 4.7, 5.0, 5.1, 5.6, 6.2, 6.8, 7.5, 8.2, 9.1

Available in multiples of

0.01 \times , 0.1 \times , 1 \times , 10 \times , 100 \times , 1k \times , 10k \times , 100k \times , 1Meg \times , 10Meg \times , 100Meg \times

5% tolerance: \$0.05 each,

Surface Mount Inductors:**Base Factors:**

1.0, 1.2, 1.5, 1.8, 2.2, 2.7, 3.3, 3.9, 4.7, 5.6, 6.8, 8.2

Available in multiples of:

0.1 μ H, 1.0 μ H, 10.0 μ H, and 100 μ H .

20 % tolerance: \$0.68 each,

10 % tolerance: \$1.13 each

Also available in multiples of:

1.0 nH, 10 nH, 100 nH

20 % tolerance: \$0.75 each,

10 % tolerance: \$0.95 each

Surface Mount Op-Amp:

Micropower Precision Op-Amp: \$1.85 each (single op-amp/chip)

Micropower Precision Dual Op-Amp: \$3.50 each (two op-amps/chip)

Micropower Precision Quad Op-Amp: \$6.12 each (four op-amps/chip)