A DIGITAL LOUDSPEAKER
EQUALIZATION TECHNIQUE

A graduate project submitted in partial fulfillment of the requirements
For the degree of Master of Science in
Electrical Engineering

By

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DEDICATION

To my wife, Anna Pomerantz:

Thank you for your support, encouragement, and patience.

I could not have finished this journey without you.
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LIST OF SYMBOLS

$B'\ell$  Transducer motor strength measured in tesla-meters

$C_{ms}$  Transducer mechanical compliance measured in $\mu$m/N

$f_3$  System Minus three decibel half-power frequency (Hz)

$f_s$  Resonance frequency of the transducer (Hz)

$L_{cvc}$  Transducer voice-coil inductance in mH

$M_{ms}$  Acoustic mass of the transducer diaphragm assembly including the air load measured in grams

$Q$  Ratio of reactance to resistance in series circuit or reactance to resistance in parallel circuit

$Q_{es}$  Transducer electrical $Q$

$Q_{ms}$  Transducer mechanical $Q$

$Q_{ts}$  Total $Q$ of transducer (woofer) considering all transducer resistances

$R_e$  DC resistance of the transducer voice-coil measured in ohms

$S_d$  Effective surface area of the driver cone measured in cm$^2$

$V_{as}$  Volume of air having the same acoustic compliance as the transducer suspension measured in liters

$X_{\max}$  Peak linear displacement of transducer cone measured in mm
ABSTRACT

A DIGITAL LOUDSPEAKER EQUALIZATION TECHNIQUE

By

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Master of Science in Electrical Engineering

This project illustrates a method of implementing a digitally equalized loudspeaker. A two-way loudspeaker using off-the-shelf transducers will be designed using the best construction methods derived from the literature. This technique will allow the author the optimum frequency response using the available components. The Harman Audio Test System (HATS) will be used to obtain acoustic measurements in a 4pi anechoic chamber. These measurements and their processed results using HATS will characterize the acoustic performance of the un-equalized loudspeaker. HATS will then be used to determine the target functions needed to equalize the loudspeaker for flat on-axis frequency response. Vectors of the target functions will be exported to MATLAB where they are used to generate the coefficients of two 128-point FIR filters. The filter coefficients will then be implemented into a DSP evaluation board. An IIR implementation using values derived from HATS will be created with tools built into the DSP software. A listening comparison can then be performed using FIR or IIR equalization versus no DSP, or pass-thru, for subjective evaluation. Anechoic measurements will be used to verify the loudspeaker has an improved frequency response based upon the current psychoacoustic literature. Finally, further applications and existing products with similar technologies will be discussed.
INTRODUCTION

The origin of the loudspeaker can be traced back to Alexander Graham Bell and the telephone [1]. The first binaural sound reproduction (meaning true stereo sound) dates back over a century to 1881, when soon after the invention of the telephone, an experiment was conducted at the Paris Opera demonstrating the use of two separate channels. These consisted each of a telephone transmitter, an interconnecting wire and a telephone receiver [2]. This primitive version of a stereo loudspeaker system was far from today’s modern designs.

It was not until 1923 and the initiation of regular radio broadcasting by the British Broadcasting Company that high quality sound reproduction actually began [3]. Higher fidelity was demanded and improved versions of Bell’s original transducer resulted in the moving-iron loudspeaker [3]. This speaker consisted of a cone driven at its apex by an iron armature moving in a magnetic field [3]. The speaker’s name is derived from the fact that part of the iron of the magnetic circuit moves. By 1925, General Electric engineers Chester W. Rice and Edward W. Kellogg presented the first moving coil loudspeaker. Their paper entitled “Notes on the Development of a New Type of Hornless Loud Speaker” [4] is the basis for the modern dynamic loudspeaker.

Still, there was much more work to be done. Research was devoted to both the objective and subjective (or psychoacoustic) measurement of loudspeakers. Advancements in acoustics led to a better understanding of how to equalize a loudspeaker using the inherent characteristics of the transducers, enclosure materials, box shape and size, and by the use of absorbent filling materials. By the early 1960’s, the electro-
mechanical analogies and basic design characteristics of moving coil direct radiators were well understood. As demonstrated in his influential paper entitled “Loudspeakers” [5], Olson and his contemporaries had substantially developed this area of research. At the same time, inventors were beginning to understand the need for multiple transducers to achieve true full range (20Hz – 20,000Hz) designs. These designs would require electrical networks to separate the various frequency ranges of each transducer. Two-way, three-way and even four-way designs were created using a variety of techniques.

Unfortunately, the wide manufacturing tolerances of the transducers produced during the 1960s often reduced the effectiveness of the crossover networks. Since the crossover frequency is usually located in the area of the sound spectrum most sensitive to human hearing, networks needed to be adjusted to individual transducers. This meant elaborate networks had to be devised that allowed user adjustable components. Fortunately, modern production processes and a better understanding of the loudspeaker system have enabled more systematic design approaches [6]. Still these analog methods of adjusting the frequency response were limited. Introducing additional components for better control adds cost and changes the overall response of the system. The result is a lot of trial-and-error in design and manufacturers being forced to choose between best response and lowest cost. Design tools continued to progress with the creation of standard design characteristics for system design called the Thiele/Small Parameters [7]. Further work led to improved analog networks, including the famed Linkwitz-Riley crossovers [8] [9]. Still, even experienced designers and engineers do not agree on the best characteristics for a loudspeaker [10]. The ability for more control was needed. Into this problem we introduce the digital (DSP) crossover.
Digital signal processing allows far more control of the system. A well-designed DSP does not load the system it is being used to filter. Thus, the effects of adding a digital filter in pass-thru mode are negligible. This fact alone makes DSP loudspeakers much easier to design. In addition, DSP provides for unlimited order filters and the ability to completely tailor the response of the system. While this will not make a poorly designed speaker great, it will allow a well-designed loudspeaker to be significantly improved.

While the theories and algorithms for digital filtering were well established by the 1970s, inadequate processing power and the high cost of DSP ICs did not allow the technology to be widely available. Most DSP power was reserved for Audio/Video receivers and post processors. These devices would allow for the decoding needed in THX, Dolby, and other digital surround algorithms [11]. The increasing power of DSP ICs and the reduction in cost achieved in the first decade of the 2000s have now enabled them to be included in speakers. As speakers are increasingly self-powered, the addition of DSPs for system equalization and other enhanced features is becoming commonplace.

This paper will show the design of a loudspeaker utilizing modern techniques. The process will begin by following the best speaker box practices in order to produce a design that has good characteristics for digital equalization. Measurements will be taken showing the frequency response of the system without the crossover. Crossover target functions will be created for best equalization and then implemented using a finite impulse response (FIR) filter generated in MATLAB. The FIR coefficients will be programmed into a DSP and measurements for the filtered system will be taken. A second filter using the infinite impulse response (IIR) method will also be made for
comparison purposes. The paper will discuss how the filtered speaker meets the objective of improved fidelity through digital filtering based upon the current knowledge of psychoacoustics. The as-built loudspeakers will also be available to the committee for subjective listening. Finally, the future applications of this technology will be considered.
I. LOUDSPEAKER DESIGN AND CONSTRUCTION

A number of important characteristics must be considered when starting a loudspeaker design. Typical factors that may be considered include: frequency response (amplitude response), sound power, directivity index, harmonic distortion, spurious noises and inter-modulation distortion, frequency shift including FM distortion, dynamic range compression (power compression), transient distortion, phase distortion, group delay, electro-acoustic efficiency, power handling capacity (or loudness), constancy of performance, size (and appearance), cost, and design time [12]. The order of importance for these priorities differs by design and has changed over the years. While the purpose of this report is to show how to equalize a speaker using DSP, there are important first steps that cannot be ignored. In order to successfully equalize a loudspeaker (using any method), it must have a basic design that will allow for proper alignment. This means the box size and tuning must produce a realizable frequency response that is useful for further equalization with the crossover network.

Most designs begin with the selection of the transducers. For this design, the author was restricted to using standard components that can be easily and affordably purchased. The transducers selected were a generously donated set of JBL 560 GTi Series speakers. This tweeter and woofer set have been designed by JBL engineers to perform well together in a correctly tuned and equalized system. The GTi 560 Series is a competition speaker system meant for automotive use. It consists of two 5 inch woofers and two 1 inch tweeters. Many of the issues mentioned above were already solved (or determined) by the choice of transducers. Transducer design and selection is a very important part of any speaker system. The design and characteristics of transducers is
Figure 1: GTi 560 Woofer [13]

quite involved and beyond the scope of this report. A good starting point for further information about transducer design can be found in Chapter 1 of the *Loudspeaker Design Cookbook* [14]. Appendix A shows the Thiele/Small specifications for the GTi 560 woofer. These were used to determine the size of the box required to meet the desired frequency specifications. The author decided to limit the target low frequency response of the design to -3dB at 60 Hz. This was chosen to prevent distortion of the woofer at low frequencies. In order to extend the low frequency response to achieve this specification, the speaker was built as a vented box (more commonly known as ported).

A vented box loudspeaker incorporates an opening which allows air to move in and out of the enclosure in response to pressure variations within the enclosure. The technique exploits the Helmholtz resonance of the box and port system. The design of

2. Nomex Spider: Provides linear force in both movement directions.
4. Copper Polepiece Cap: Provides linear inductance over the full range of focused voice-coil travel for reduced intermodulation distortion. Provides crystal-clear vocals and midrange, even during heavy bass signals.
5. Polished and Flared Polepiece Vent: Provides a low-velocity inlet and outlet for the movement of air in and out of the motor structure. Minimizes distortion from mechanical noise.
6. Neodymium Magnet: Provides high flux density. Also allows more room for larger steel motor components to provide critical heat sink mass for the voice coil.
7. Vented Gap Cooling Ports: Provide movement of air over the voice coil for superior power handling.
8. Flux Stabilization Ring: Provides global stabilization of the static magnetic field and works with the copper cap to minimize coil inductance during inward movement of the voice coil.
9. Voice Coil: Long, over-hung 2" diameter, aluminum edge-wound voice coil provides high excursion for improved low-frequency capability. Reduces distortion at low frequencies and high input power.
10. Vented Voice Coil Former: Minimizes distortion from mechanical noise.
11. Screw-Down Terminal: Ensures reliable high-quality connections.

...
vented enclosures is covered in detail in papers by Thiele and Small [15] [16]. The basic process required for alignment as used for this design will be reproduced here. The steps follow the procedure described in the *Loudspeaker Design Cookbook* pgs 56-78 [14] and Small [16] with the exception that the project was started with a predetermined box volume.

1. Select an appropriate woofer. The GTi 560 Woofer has a rated $Q_{ts}$ of 0.34 which satisfies the requirement of $Q_{ts} = 0.2 - 0.5$. The measured value changed to 0.45, which is still suitable for use in this project.

2. Typically, alignment would be chosen next. For this project, the box volume was set at $V_b = 0.38 \text{ ft}^3 (10.8 \text{ liters})$. This was done since choosing the alignment first could lead to the requirement of producing different sized boxes to arrive at the correct tuning. The value was also based upon the guidance of experienced JBL engineers.

3. Choose an alignment. A flat Quasi-Third Order (QB3) alignment was selected. It is the most common vented alignment as it yields a smaller box size and lower $f_3$ for a given driver $Q_{ts}$ [14].

4. Determine the $B\ell, f_s, Q_{ts}, V_{as}, X_{max}, S_d$, and $V_d$ Thiele/Small parameters for the woofer. All of these are given in the woofer data sheet except for $V_d$ which is calculated as:

$$V_d = S_d \times X_{max} = (8.67 \times 10^{-3} \text{ m}^2) \times (4.87 \times 10^{-3} \text{ m}) \equiv 4.22 \times 10^{-5} \text{ m}^3$$
Note that values were adjusted for the actual driver as measured in the lab. For more information on loudspeaker testing see chapter eight of [14]. The adjusted values are shown in Appendix A.

5. Assume leakage loss, $Q_L$, is equal to 7.

6. Use alignment chart per Small [16] with measured values of $Q_{ts}$ and calculated value for $\alpha$. See Appendix B for MATLAB code showing the calculations and the results. The results of plotting the derived values on the alignment chart are shown below.

![Alignment Chart for Vented-Box Systems](image)

**Figure 2:** Alignment Chart for Vented-Box Systems with $Q_L = 7$ [16]

Shown with plot of project values in red
It can be seen from Appendix B that the calculated value of $\alpha \equiv 0.66$. We find from Figure 2 that $h \equiv 0.87$ and $f_3/f_s \equiv 0.77$. We also find the calculated value of the $-3$dB frequency for the design to be approximately 57Hz (See Appendix B).

7. Determine the port length and diameter. Using the calculations in Appendix B we find the important values for the port. These are: minimum port diameter of just under 2 inches and port length of approximately 4 inches. The port, the PSP2 made by Precision Sound Products (http://www.psp-inc.com), has a one-inch flare on each end to prevent air noise. The general guideline for the effective port length is to reduce the calculated length by half the flare radius on each end (see Figure 3). Thus, the designed port length for the project is 5 inches (see Figure 4) resulting in an effective length of our calculated 4 inches.

![Figure 3: Effective Port Length for a Flared Port](image)
Figure 4: Port Design for the Project showing a length of 5 inches

The alignment of the project is now complete. There are still many more items that could be assessed to make an ideal loudspeaker. For this project, the other items considered include: diffraction, transducer placement, directivity, and enclosure structural design.

Diffraction is the process by which the change of direction of sound takes place [17]. Olson demonstrated that the exterior of the cabinet influences the response of the loudspeaker. This change in response is due to the diffraction effects produced by the surface contours of the cabinet [18]. Olson measured the performance of a great many cabinet shapes loaded with a direct radiator loudspeaker. The results showed that reducing sharp boundaries on the front portion of the speaker reduces variation in response. Thus, for this project, all corners on the loudspeaker are well rounded with no
sharp edges on the baffle. The next issue, transducer placement, is important to prevent phase and directivity problems. In order to minimize this issue, the transducers were spaced as closely together as possible. In addition, the tweeter is mounted close to the baffle top at a distance different than its distance to the left and right sides of the box. The purpose is to vary the distance to the diffracting edges to create a more random phase relationship between the primary and diffracted sound waves [18]. In addition, a waveguide included with the GTi 560 series was used for mounting of the tweeter. The purpose of the waveguide is to acoustically load the tweeter and thereby narrow the directivity pattern. The goal is to focus the acoustical energy of the tweeter and thereby match the directivity of the woofer at the crossover point. More info on directivity related to transducer spacing can be found in reference [19]. The final issue considered in the cabinet design was the construction. For ease of manufacturing and consistency of wall density, ¾ inch medium density fiberboard (MDF) was used for the walls and brace. The front baffle was made extra thick (1 inch) to further stiffen the unit against the first cabinet resonance. A brace was utilized to provide additional rigidity. The cabinet walls were lined with one-inch thick sheets of fiberglass to reduce sound radiation and provide improved damping. The cabinet drawings can be seen in Appendix C. See reference [20] for more info on loudspeaker cabinet materials and construction.
II. MEASUREMENTS

Measurements were taken of the raw (un-equalized) loudspeaker using the 10,000 cubic foot JBL 4pi anechoic chamber. This chamber has large 4-foot wedges, which allow for echo-less measurements from 60 Hz to over 20kHz (+/- 0.5dB, 1/20\textsuperscript{th} octave). Additional calibration enables the chamber to be accurate down to 20Hz (+/- 0.5dB, 1/10\textsuperscript{th} octave) [21]. The speaker was placed 2m from the measurement microphone on a platform that rotates in 10\textdegree intervals from on-axis (0\textdegree degrees) to 350\textdegree. The height of the rotating platform is adjusted so the reference axis (or midpoint between the tweeter and the woofer) is centered at the correct measurement location. This same process is applied for both the horizontal and vertical axes. Thus, 72 measurements are taken in total. Figures 5 and 6 show the horizontal and vertical setup of the speaker in the chamber.
The test results have been auto-adjusted to the reference standard of 1 meter so the sensitivity can be read directly [23]. Sensitivity is a measure of the acoustic sound-pressure level produced by a stated electrical input voltage [6]. The typical standard is to measure at a distance of 1 meter with an input reference voltage of 2.83V. Loudspeaker standards specify that measurements should be made in the far field, which is defined as the point where sound level decreases at -6 dB per doubling of the distance. For most loudspeakers, a distance of 2 meters is adequate for this requirement. After taking each measurement at 2m, the sound level that would be expected from a point source at 1m is then calculated [23]. This is the reason for the auto-adjustment of the levels. Unfortunately, the need for separate amplification of the tweeter and woofer meant the tests had to be performed with an un-calibrated stereo reference amplifier. A Proceed Amp 2 was used instead of the reference Lexicon 501 mono-block amplifier typically used in the laboratory. Thus, the SPL levels in this report are reference values only. For
a true comparison, the raw speaker measurements had to be acquired with the DSP board in-line before the amplifier and set to pass-thru mode. We will forego an in-depth discussion of the DSP board until Section III.

The interface for taking these measurements is called HATS or the Harman Audio Test System. The HATS software is a powerful sixteen channel audio analyzer that is optimized for measuring loudspeaker and loudspeaker-room systems [22]. The HATS analyzer utilizes a BSS BLU-32 hardware I/O for the A/D and D/A used in taking the measurements. HATS has many built-in stimuli for audio testing and can be configured with the BLU-32 for a diverse set of applications. More about the software can be read in the HATS User Manual [22]. Various templates can be made in HATS for standard acoustical tests. For the 72 measurements described above and their corresponding analysis, an engineering template called the Spin-O-Rama was utilized. See Appendix D for a screenshot of the Spin-O-Rama user interface. This template runs the test and provides objective loudspeaker output data. The horizontal and vertical must be set up and run individually but the 36 measurements for each are automated.

The horizontal and vertical anechoic speaker measurements at 0°, 10°, 20°, 30°, 60°, 90°, 120°, and 180° are shown below. From the anechoic measurement curves, we can see a clear discontinuity at approximately 1200 Hz. This is where the woofer response drops before the rise of the tweeter response. There is a clear level difference between the tweeter and the woofer of over 10 dB. The large peak at 2kHz in the horizontal response is caused by the acoustic loading of the waveguide.
Figure 7: Horizontal Loudspeaker Amplitude Frequency Response (No Crossover)

Figure 8: Vertical Loudspeaker Amplitude Frequency Response (No Crossover)
HATS also allows us to generate other key data used in analyzing the speaker frequency response. These include the listening window, early (or first) reflections, sound power, and the directivity index (DI). The meaning of these terms and their curves will now be explained:

Listening Window: Listening Window refers to the spatial average of the nine frequency responses on the ±10° vertical and ±30° horizontal angular range [23]. This area of the speaker is the most common listening area. The spatial averaging attenuates small acoustical interferences that have little effect on the overall sound quality and reveals resonances which can be audible.

Early (or First) Reflections: Sounds that have been reflected only once on their way to the listener. The curve is an estimate of all single-bounce, first reflections in a typical listening room. Being that it is a spatial average, a bump that appears in this curve and also appears in other curves provides clear evidence of a resonance [23]. For more information about the research behind this curve and its calculation see [21].

Sound Power: Sound power is meant to represent all the sounds arriving at the listening position [23]. As such, it is a weighted average of all the horizontal and vertical measurements. The weighting is applied to make the results representative of the human listening experience. The sound power is a measure of the total acoustical energy radiating through an imaginary spherical surface with radius equal to the measurement distance [23].
Directivity Index: The directivity index is defined as the difference between the on-axis curve and the sound-power curve. The on-axis curve often has diffraction artifacts related to the shape of the cabinet or transducers that do not appear in other measurements. In order to prevent these artifacts from affecting the curve, the definition has been redefined to make the DI (directivity index) the difference between the listening window and the sound power curves. This curve is a measure of the degree of forward bias (or directivity) in the sound radiated from the loudspeaker [23].

Figure 9: Key Loudspeaker Data for Un-equalized Loudspeaker including: On-Axis Amplitude Response, Listening Window, First Reflections, Sound Power, Directivity Index (DI)
Figure 9 shows the key measured and calculated metrics for the project loudspeaker. (Note that Appendix E shows the 0°, 10°, 20°, 30°, 60°, 90°, 120°, and 180° anechoic speaker measurements and their calculated metrics in larger and separated graphs for ease of viewing.) Here again, we see the large discontinuities observed in the individual responses. This result makes sense as these individual responses are used in the calculated curves. It is not possible to objectively quantify the sound character of a loudspeaker in a single measurement. Thus, the current best method uses many curves and the relationship between these curves to define the response. Section IV will discuss this topic in greater detail. What we can understand at this point is the need for amplitude equalization of the speaker based upon the differences in level between the tweeter and the woofer and the loss of audio at 1200 Hz. It should be noted that items including phase response, non-linear distortion, and other problems could also be measured and corrected. Since the human ear has shown through testing to be insensitive to speaker phase variations, this problem will not be considered in depth [23]. Nonlinear distortion will also not be considered as its audibility is affected by the type of distortion [25] [26]. While it is possible to correct for these issues in DSP, they will be considered outside the scope of this project.
III. EQUALIZATION PROCEDURE

For this part of the project we again start by turning to HATs. HATs has the capability to apply virtual transfer functions to the measured and calculated data. Equalization (EQ) curves can be applied directly in HATs and the results can be seen in real time. This is a very powerful aid to the would-be loudspeaker designer as it allows for trial-and-error without the need of additional measurements. Based upon the results of the raw curves generated for the project speaker, we can conclude that adjustments are needed to improve the overall amplitude frequency response.

The first question that needs to be asked is: “What is the ideal amplitude response of a loudspeaker?” There is no one answer to this question. Many papers have been written on this topic and as yet there is no definitive answer. Based upon the research of Toole and Olive [26] [27] [28] [29], a high correlation is shown between listener preference and loudspeakers with flat on-axis amplitude. Thus, for this project, flat on-axis amplitude was established as the optimum speaker amplitude response. With this goal in mind, HATs can now be utilized to optimize the speaker’s response. The Spin-O-Rama template allows the use of a global EQ, a high frequency virtual transfer function, and a low frequency virtual transfer function to create the crossover and apply additional EQ.

The process of equalizing a speaker in HATs mimics the traditional trial and error methods. It requires an understanding of filters, loudspeakers, and their interactions. As stated above, the great advantage of using HATs is the ability to make adjustments and see the results in real time without the need for physically rebuilding the crossover or
taking additional measurements. Still, there is no substitute for experience since the great flexibility of the tool can allow for endless possibilities in design. Knowing what filters to apply at the starting point is essential. Reviewing the raw data in Section II, we found the approximate crossover frequency to be 1200 Hz. At this point the Low Frequency Network Virtual Transfer Function (LF Network VTF) window was activated and a third order maximally flat (or Butterworth [14]) low pass filter was created with center frequency of 1300 Hz. Next, the High Frequency Network Virtual Transfer Function (HF Network VTF) window was activated and a second order Butterworth High Pass filter was created with center frequency of 3000 Hz. The LF and HF Network VTF windows are shown in Figures 10 and 11. Global EQ was not utilized for this project.

![Virtual EQ - 219: LF Network VTF](image)

**Figure 10:** The HATS Low Frequency Network Virtual Transfer Function Showing the Data of the Low Pass Filter

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For low-pass and high-pass filtering, HATS can apply Butterworth or Linkwitz-Riley [8][9] filters. The crossover frequencies and types used in this project were chosen by trial-and-error to produce the best overall flat amplitude frequency response. As can be seen in Figures 10 and 11, additional parametric EQs were applied to improve the overall response. Table 1 shows all the filters applied including these additional EQs and their data. The purpose of the EQs was to prevent peaks or dips apparent in the predicted response curves. The result of applying these curves on the important loudspeaker curves is shown in Figure 12. Here we can see the on-axis has been significantly improved from the raw data. We also find the other curves are smooth which has been shown to correlate well with listener preferences [23]. Remember that these curves including the
revised on-axis response are predicted values that are yet to be measured for verification in the anechoic chamber.

<table>
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<th>Filter Type</th>
<th>Center Freq (Hz)</th>
<th>Filter Gain (dB)</th>
<th>Bandwidth (Octaves)</th>
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<td>3rd Order Butterworth</td>
<td>1300</td>
<td>0</td>
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<td></td>
<td>Parametric</td>
<td>N/A</td>
<td>200</td>
<td>1.5</td>
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<td></td>
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<td>N/A</td>
<td>900</td>
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<td>3000</td>
<td>-7</td>
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<td></td>
<td>Parametric</td>
<td>N/A</td>
<td>2100</td>
<td>-2</td>
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<td></td>
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<td>N/A</td>
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Table 1: Virtual EQ Filters Applied in HATS

Figure 12: HATS predicted speaker response curves with added virtual EQs
The next step is to output the results of the virtual transfer functions for use in creating the FIR coefficients in MATLAB. This was accomplished by exporting the HF VTF Target and LF VTF Target Rows of the Spin-O-Rama worksheet. Figures 13 and 14 show the LF and HF virtual target curves, which are composites of the EQs shown in Table 1. The result of exporting the curves is two text files that contain vectors of the frequency, magnitude, and phase of the target curves. These files could not be included in this report due to their length. The files were then imported into MATLAB. A MATLAB program was written to provide the FIR coefficients of both the 128-point low frequency and 128-point high frequency filters. The magnitude frequency response of the resulting FIR filters should match those shown in Figures 13 and 14.

Figure 13: Low Frequency EQ Target Curve output from HATS
Appendix F shows the code for the MATLAB program hats2fir.m. The code has comments that describe each step in more detail. A general synopsis for the function of this code is to provide the \( b_k \) filter coefficients for the general FIR filter of the form:

\[
y(n) = b_0 x(n) + b_1 x(n-1) + \cdots + b_{M-1} x(n-M+1) \\
= \sum_{k=0}^{M-1} b_k x(n-k)
\]

The filter was implemented by first importing the HATS generated text files into a MATLAB data file. The data was then read into frequency, magnitude, and phase vectors. This data was then re-sampled and interpolated using the \texttt{interp1} function of MATLAB. The complex frequency response was then created and the inverse Fourier
transform was taken using the MATLAB function \texttt{ifft}. Only the real part needed to be considered. Note that the result of this technique produces the impulse response, which is the same as the FIR coefficients [30]. The resulting impulse responses were then plotted and are shown below in Figures 15 and 16. The .DAT files with FIR coefficients can be seen in Appendix F.

![Impulse Response Graph]

\textbf{Figure 15:} Low Frequency FIR filter Impulse Response
Figure 16: High Frequency FIR filter Impulse Response

The final step is to implement the FIR coefficients into the DSP board for filtering of the loudspeaker. Figure 17 shows the generalized system block diagram. The DSP integrated circuit (IC) utilized for the project is the Cirrus Logic CS47048C. A circuit board was required to gain access and control the CS47048C IC. Since designing the circuit from scratch was considered beyond the scope of the project, the CDB47XXX evaluation board was used to implement the filters. This board has multiple analog and digital I/Os to the onboard CS47048C DSP IC. In a real system, only the DSP would be utilized within the architecture of the product. Cirrus provides a software interface to the
IC called the DSP Composer. The software comes with built-in signal processing modules, and the visual interface provides for ease of design.

![Diagram](image)

**Figure 17: Project Loudspeaker System Block Diagram**

Building a project within DSP Composer requires certain component blocks to be placed within the system. Once these blocks have been included, the custom portion of the design may be implemented within the Custom Post-Processing Module (or Custom PPM) block. For the project, the custom FIR primitive was used for creation of both the woofer and tweeter equalization. Once the component blocks are properly placed and the system is correctly routed, all that is necessary for implementation of the FIRs is to point the primitive to the correct file location for the FIR coefficients. The project can then be built and tested. Figures 18 and 19 show the system design within DSP Composer for the FIR filter design.

For comparison purposes, a second version of the project was built using the IIR method. For this method, the EQs shown in Table 1 were created using the built-in primitives for low pass, high pass, and parametric filtering. The user interface showing the implementation and three of the selection windows is shown in Figure 20. Finally, as mentioned in Section II, a pass-thru version was created for use in taking the raw measurements. This routing of the custom PPM is shown in Figure 21. Note that a 60Hz
A high pass filter was applied to the tweeters for DC protection. The completed designs are now ready for attachment to the amplifier/speaker system for measurement.

Figure 18: DSP Composer FIR System Block Diagram

Figure 19: DSP Composer FIR Custom Post Processing Module Routing
Figure 20: DSP Composer IIR Custom Post Processing Module Implementation

Figure 21: DSP Composer Pass-Thru Custom Post Processing Module Implementation
IV. RESULTS

Before testing the speaker in the anechoic chamber, a measurement was taken of the amplitude frequency response for the DSP board by itself. The measurement was taken for the pass-thru, FIR, and IIR filter implementations. Note that the pass-thru was taken before the measurements in Section 2 but is shown here for clarity. The results are shown below in Figures 22-27. We expect the pass-thru curves to be flat except for the DC filtering of the tweeter. The FIR and IIR implementations in the DSP board should produce a result that matches the target curves generated in HATS as shown in Figures 13 and 14.

Figure 22: DSP Board LF Pass-Thru showing Flat Response

Figure 23: DSP Board HF Pass-Thru showing Flat Response above 60Hz
Figure 24: DSP Board LF FIR Amplitude Response

Figure 25: DSP Board HF FIR Amplitude Response

Figure 26: DSP Board LF IIR Amplitude Response

Figure 27: DSP Board HF IIR Amplitude Response
We can see from the resulting plots that the output curves do match closely with the target curves generated in HATS. There are some differences that could be related both to the Audio Precision (AP) measurement spacing and for the FIR the interpolation of the target curve. Based upon this result, the board can now be used to filter the loudspeaker for anechoic measurements.

As shown in Figure 17, the DSP board was placed between the HATS output and the amplifier to allow for equalization. (This was also done for the raw measurements.) The same measurement technique described in Section II was again performed on the speaker with both FIR and IIR filters. The resulting important metrics including on-axis, listening window, first reflections, sound power, and directivity index are shown below in Figures 28 and 29.

![Figure 28: Key Loudspeaker Data for FIR Equalized Loudspeaker](image-url)
Figure 29: Key Loudspeaker Data for IIR Equalized Loudspeaker

The results match closely with the predicted curves from HATS (see Figure 12). In addition, the results of using the FIR and IIR on the on-axis amplitude response and other metrics are very much the same. This would seem to indicate the two implementations sound the same. Appendices G and H show the 0°, 10°, 20°, 30°, 60°, 90°, 120°, and 180° vertical and horizontal anechoic speaker measurements for the FIR and IIR implementations. It can be noted from the amplitude response measurements that the off-axis response does vary between the FIR and IIR. Subjective listening tests conducted on the two versions did confirm a difference in timbre when listening outside the listening window (the ±10° vertical and ±30° horizontal angular range).
Overall, the results show the use of the FIR and IIR DSP implementations provided improvement in the objective measurements of the speaker. The on-axis amplitude response was modified much more closely to the ideal flat amplitude. The other key metrics show smooth responses that correlate well with listener preferences [24]. Subjective listening tests also noted an overall smoother sound with enhanced midrange. Some improvements are noted. Most important is the reduction of the low-Q bump centered at 2.7 kHz. This is noticeable in the listening window, first reflections, and sound-power curves as a shallow peak and can be seen as a dip in the DI curve at the same frequency. Research by Olive and Toole noted that these types of resonances can be most noticeable in the sound quality of a loudspeaker [30]. Preventing the occurrence of these low-Q bumps can reduce coloration and provide the loudspeaker with a natural and open sound. It should be noted that the testing and implementation of this project was undertaken with the assumption that both loudspeakers have the same Thiele/Small parameters and therefore would have identical amplitude response. In reality, this can never be the case due to variation within the transducers. During testing, both speakers were measured on-axis for a comparison and the result was very close (within ±1 dB). Unfortunately, this data was not saved and therefore is not available for this report.

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V. APPLICATIONS

As the cost of DSP ICs decrease and loudspeakers become increasingly self-powered, this form of equalization is becoming attractive. In addition to the reduction in component cost and size there is also a cost savings in terms of engineering. For product development purposes, this technique allows the ability to test new EQs on the spot without the need of rebuilding the unit. It also provides the flexibility of infinite filtering to easily control problems that formerly would have required time-consuming engineering solutions. As mentioned in Section IV, most loudspeaker designs assume the variation in transducers to be within a certain range. This assumption is not always valid as the Thiele/Small values can vary greatly between builds. Using this method, each individual loudspeaker can be calibrated on the line to the reference standard. The frequency response of the unit can be measured and the impulse response modified to assure the most exact match. This would allow for highly precise matching between loudspeaker pairs. In addition, the powerful DSPs available today would allow for custom tailored EQs and/or control of room modes by the end-user. This increased flexibility would allow consumers to tailor their loudspeaker to personal taste, or remove unwanted frequencies caused by problem room acoustics or poor recording. In today’s market, the ability to provide a customizable listening experience can set a product apart from the competition.

Some examples of products that are already implementing this forward-looking technology include:
• The Genelec 8260A loudspeaker. A bookshelf speaker that is ideal for use as a professional monitor. The user can modify the speaker’s frequency response through a series of selection switches on the rear of the unit. Alternately, the speaker can be controlled through the Genelec Loudspeaker Manager software. The speaker may be calibrated for best in-room response using the included microphone and AutoCal™ software [31].

• The Meridian DSP8000 Digital Loudspeaker. This product is an example of an audiophile quality DSP equalized loudspeaker. These speakers allow for digital connection to prevent possibility of noise-induced hum from the analog cables. The built-in DSP allows for custom crossovers with linear phase, steep slopes, and time delay compensation. The speaker allows for beam steering volume control that is adjusted to the listener’s horizontal and vertical position [32].

• The JBL OnBeat™ Loudspeaker Dock. This dock is for use with the iPod or iPad. It utilizes DSP equalization to provide a richly detailed 360-degree soundstage [33]. The sound is quite impressive especially for the product’s size, price, and uniquely stylized appearance (which can often hinder great sound capability).
VI. CONCLUSION

This project demonstrates the design and construction of a loudspeaker equalized using a digital signal processing technique. The entire process was covered, including best loudspeaker tuning and construction practices, measurements, equalization (both FIR and IIR methods) and results. The project has been a success since the equalized speaker is a significant improvement both objectively and subjectively over the un-equalized design. The process illustrated in this paper closely matches the most current techniques employed in industry. This paper notes additional areas of research that could be explored including reduction of noise and distortion, improved phase linearity, in-room response control, improved directivity control, and others. During product development, these areas would also need further exploration and consideration as part of the equalization process.
REFERENCES


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<th>Measured Values</th>
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<td>$7.6$</td>
<td>$7.22$</td>
</tr>
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<td>DC Resistance of the Voice Coil (ohms)</td>
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<td>$3.7$</td>
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<tr>
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<td>$14.84$</td>
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<td>Effective Projected Diaphragm Area (cm$^2$)</td>
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<td>$86.6$</td>
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<td>Driver Mechanical Compliance ($\mu$m/N)</td>
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<td>$7.06$</td>
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<td>$74$</td>
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<td>$0.45$</td>
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<td>Distance the Voicecoil can travel maintaining constant # of turns in the gap (mm)</td>
<td>$4.87$</td>
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APPENDIX B

MATLAB LOUDSPEAKER CALCULATIONS

%California State University, Northridge
%MSEE Project
%A Digital Loudspeaker Equalization Technique
%By Colby Buddelmeyer
%Dec 2011

%file: Loudspeaker_Calcs.m

%Program of calculations used to determine:
%Lv port length, f3 (-3dB half power frequency), h (tuning ratio)
%alpha (system compliance ratio), dv (min port diameter)

%inputs include mfgr given transducer values
%(which are modified for measured DC Voicecoil resistance)
%and chosen box size and chosen port diameter

close all
clear all
clc

format short

%reference values
$c = 345; %velocity of sound in air (m/s)$

%Mfg supplied T/S Parameters
$BL_{mfgr} = 7.6; %Driver Motor Strength (Tesla-Meters)$
$Re_{mfgr} = 3.07; %DC Voicecoil resistance given by mfgr$
$fs = 74; %Driver Free Air Resonance (Hz)$
$Qts_{mfgr} = 0.34; %Driver Total Q$
$Qms = 5.49; %Driver Mechanical Q$
$Qes_{mfgr} = 0.37; %Driver Electrical Q$
$Vas\_liters = 7.06; %Volume of Air Equal to the Driver Compliance, CMS (liters)$
$Vas = Vas\_liters*10^(-3) %convert Vas to meters cubed$
$Xmax = 4.87e-3; %amount of voicecoil overhang (m)$
$Sd = 8.66e-3; %Effective Projected Diaphragm Area (m^2)$
$Cms = 3.12e-4; %Driver Mechanical Compliance (m/N)$
%-------------------------------------------------------------
%adjusted values for measured values
BL = 7.22;
Re = 3.7; % Measured DC Voicecoil resistance
Qes = Re/(BL^2*Cms*fs*2*pi) % Driver Electrical Q adjusted for real Re
Qts = 1/((1/Qms)+(1/Qes)) % Driver Mechanical Q adjusted for real Re

%-------------------------------------------------------------
% Selected values:
Vb = 658; % liters selected (658in^3 = 10.76 liters)
r = 1; % choose 2in diameter port, radius = 1in

%-------------------------------------------------------------
% calculated values
Vd = (Sd)*Xmax % Displacement volume (m^3)
alpha = Vas_liters/(Vb*(0.016387)) % system compliance ratio
h = 0.87; % tuning ratio from Figure 2
fb = h*fs % box tuning frequency (Hz)
Lv = ((1.463e7*r^2)/((fb^2)*Vb))-1.463*r % port length in inches
f3_fs = 0.77; % f3/fs ratio from Figure 2
f3 = f3_fs*fs % -3dB frequency
dv = 39.37*((411.25*Vd)/sqrt(fb))^(.5) % minimum port diameter
File Output:

Qes =
0.4893

Qts =
0.4492

Vd =
4.2174e-005

alpha =
0.6548

fb =
64.380

Lv =
3.9013

fB =
56.9800

dv =
1.8304
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APPENDIX E

UNEQUALIZED SPEAKER MEASUREMENTS

ON AXIS

10 DEGREES HORIZONTAL
20 DEGREES HORIZONTAL

30 DEGREES HORIZONTAL
60 DEGREES HORIZONTAL

90 DEGREES HORIZONTAL
0 DEGREES VERTICAL

10 DEGREES VERTICAL
120 DEGREES VERTICAL

180 DEGREES VERTICAL
APPENDIX F

MATLAB HATS2FIR.M CODE AND .DAT OUTPUT FILES

%hats2fir.m
%ECE698C - Fall 2011
%Master's Project
%Student: Colby Buddelmeyer

%Create HF Target FIR Coefficients
close all
clear all
clc

fs=48000;    % fs
ny=fs/2;     % nyquist
N=2^9;       % block size
N2=2^7;      % FIR crop length
f1=(1:N)/N*ny;

% hats data
pathin = 'C:\target_curves\';   %path for input txt file
filename= 'HF VTF Target.txt'; %filename
data1 = importdata([pathin filename],'	',5); %5 header lines
datatemp = data1.data;

f2=datatemp(:,1); % freq (Hz)
A2=datatemp(:,2); % Mag (dB)
phi2=datatemp(:,3); % phase (deg)

% resample
A1=interp1(f2,A2,f1);
phi1=interp1(f2,phi2,f1);

% -> H
[Hre Him] = pol2cart( pi.*phi1./180, 10.^A1./20 ); %polar -> rectangular
H=complex(Hre,Him); %cmplx
H=[H;conj(flipud(H(2:N-1)))];

% -> generate HF impulse response (FIR coefficients)
h=real(ifft(H));

% check plots
figure
stem(h(1:N2))
grid
title('HF Impulse Response')
xlabel('Sample')
ylabel('Magnitude')

% export FIR coefficients for Cirrus DSP
filename = 'HF_VTF_Target.dat';
fid = fopen(filename,'w');
for i=1:N2
    fprintf(fid,'%1.10f
',h(i));
end
fclose(fid);

%-------------------------------------------------------------%
%-------------------------------------------------------------%
%Create LF Target FIR Coefficients
%-------------------------------------------------------------%
clear all
clc

fs=48000;        % fs
ny=fs/2;         % nyquist
N=2^9;          % block size
N2=2^7;         % FIR crop length
fl=(1:N)/N*ny;

% hats data
pathin = 'C:\target_curves\';   %path for input txt file
filename = 'LF_VTF_Target.txt'; %filename

data1 = importdata([pathin filename],'\t',5);   %5 header lines
datatemp = data1.data;

f2=datatemp(:,1);  % freq (Hz)
A2=datatemp(:,2);  % Mag (dB)
phi2=datatemp(:,3);  % phase (deg)

% resample
A1=interp1(f2,A2,fl);
phi1=interp1(f2,phi2,fl);

% -> H
[Hre Him] = pol2cart(pi.*phi1./180, 10.*(A1./20)); %polar -> rectangular
H=complex(Hre,Him);  %cmplx
H=[H;conj(flipud(H(2:N-1)))];
%conjugate, flip
% -> generate LF impulse response (FIR coefficients)
h=real(ifft(H));

% check plots
figure
stem(h(1:N2))
grid
title('LF Impulse Response')
xlabel('Sample')
ylabel('Magnitude')

% export FIR Coefficients for Cirrus DSP
filename = 'LF_VTF_Target.dat';
fid = fopen(filename,'w');
for i=1:N2
    fprintf(fid,'%1.10f
',h(i));
end
fprintf(fid,"
);
fclose(fid);
HATS OUTPUT TARGET CURVE .TXT FILES

LF_VTF_Target.dat (contains FIR coefficients for Low Frequency Target Curve):

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HF_VTF_Target.dat (contains FIR coefficients for High Frequency Target Curve):

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  0.0091875410   0.0043694411  -0.002561097
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APPENDIX G

FIR EQUALIZED SPEAKER MEASUREMENTS

0 DEGREES HORIZONTAL

10 DEGREES HORIZONTAL
20 DEGREES HORIZONTAL

30 DEGREES HORIZONTAL
60 DEGREES HORIZONTAL

90 DEGREES HORIZONTAL
120 DEGREES HORIZONTAL

180 DEGREES HORIZONTAL
0 DEGREES VERTICAL

10 DEGREES VERTICAL
20 DEGREES VERTICAL

30 DEGREES VERTICAL
60 DEGREES VERTICAL

90 DEGREES VERTICAL
120 DEGREES VERTICAL

180 DEGREES VERTICAL
APPENDIX H

IIR EQUALIZED SPEAKER MEASUREMENTS

0 DEGREES HORIZONTAL

10 DEGREES HORIZONTAL
20 DEGREES HORIZONTAL

30 DEGREES HORIZONTAL
60 DEGREES HORIZONTAL

90 DEGREES HORIZONTAL
120 DEGREES HORIZONTAL

180 DEGREES HORIZONTAL
0 DEGREES VERTICAL

10 DEGREES VERTICAL
20 DEGREES VERTICAL

30 DEGREES VERTICAL
60 DEGREES VERTICAL

90 DEGREES VERTICAL

81
120 DEGREES VERTICAL

180 DEGREES VERTICAL
APPENDIX J

CIRRUS LOGIC CDB47XXX DSP EVALUATION BOARD