**Audio System Analysis, Equalization and Boosting**

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**Introduction**

In the use of audio engineering having an accurate reproduction of an input signal at the output of system is a very important task to develop and perfect for various applications in our lives today, for instance high-quality sound systems. What makes this task a bit difficult is the introduction of hinderance due to distortions introduced by the audio system, such as speaker imperfections, or ambient factors such as room acoustics. These distortions really can diminish the quality of sound, which makes it less desirable and sometimes incomprehensible for anyone that chooses to listen. To be able to tackle this challenge there needs to be an effective approach to analyze and correct for distortions in the audio ensuring that the output is close to perfect replication of original input.

The primary objective of this project is to design and implement an audio speaker equaling system that can correct the distortions. Through the analysis of the transfer function of a given audio system. This project aims to develop an inverse filter that flattens out the frequency response, which removes any distortions. To start this process involves the estimation of the systems transfer function magnitude using the provided input and output audio signals. From the estimation the inverse filter can be derived and implemented to ensure that the output replicates that of the input. In this project only the magnitude is of focus thereby disregarding the phase distortions.

By correcting these distortions, the project will include the enhancement of specific audio frequencies such as bass and treble. To implement these enhancements the design process includes the simulation of the transfer functions of bass boost and treble boost with the corrected audio signal.

**Method Descriptions**

For this project we are given the task of correcting a distorted output and removing the distortions to get the original input audio. The methods to achieve such a task involves a few steps and these steps are processed to analyze and correct the system transfer function. These steps are mainly implemented using computational tools and in this project the software used is MATLAB, to design an effective inverse filter with the given data set for the file ProjectData2.mat.

**Transfer Function Estimation**

The first step in completing this project is to estimate the system transfer function. To estimate the transfer function of this audio system given the magnitude of the transfer function(|H(w)|) can be estimated by attaining power spectrum of Pyy(w) which is the ouput power spectrum as well as the power spectrum of Pxx(w) which is the input power spectrum and dividing Pyy(w) by Pxx(w) and squaring rooting it to attain the magnitude over w graph of the transfer function. To implement this in MATLAB the pwelch() function is used to retrieve the spectral estimation with parameters of Fourier transform size NFFT, data segment overlap OL and sampling frequency Fs. After finding Pyy(w) and Pxx(w) and attaining |H(w)| the power spectra of the Input and output are then visualized on a logarithmic frequency scale with the function semilogx() which will take the frequency vector F and the power spectra and convert the power values into logarithmic using the formula 10\*log10(Pxx) (for output replace Pxx with Pyy.

**Inverse Filter Design**

The system transfer function magnitude is now estimated, to be able to design the inverse filter it is simple enough to be able attain the reciprocal of the transfer function. The transfer function in terms of the variable ‘s’ is implemented in MATLAB with the use of s = tf(‘s’) which is a transfer function routine that will be used to get H(s), and to estimate H(s) there must be an estimation of the zeros and poles with their degrees based on the slopes of the transfer function magnitude graph at their respective frequencies. A first degree zero has a slope of +20dB/decade and a pole have a slope of -20dB/decade and the degree of each add another +-20dB/decade depending on it being a zero or pole respectively. Once the transfer function H(s) is estimated then we can now attain the inverse transfer function Hinv(s) but reciprocating H(s) to get 1/H(s).

Simulation and Correction

The process of correcting the output sound Y and attaining the original input sound with no distortions X, with the use of MATLAB function lsim() which performs the linear simulation of a system given the transfer function on a signal. The parameters to the lsim() function are the inverse transfer function and the output audio signal Y, which when simulated will theoretically generate the corrected output signal.

**Sound Analysis and Evaluation**

With the simulated output signal with the inverse transfer function there needs to be a validation of how effective the inverse transfer function is in eliminating the distortions of the original output. The MATLAB audiowrite() function is used to be able to save a .wav file of the signals of interest with parameters of the filename, Ys (rescaled signal vector Y/max(abs(Y))), Fs (sampling frequency (Hz)) and nbits (number of bits). Once the files are made and heard there will be a distinct difference in the original input and output as well as no difference or a large difference in the corrected output compared to the original depending on how good or bad the estimated inverse filter is. The power spectra of the signals can be shown with semilogx() function to show the differences between the signal’s frequency ranges and their magnitudes. There also is need of showing the time waveforms of the three signals so the use of plot(T, signal) is sufficient of being used to plot the time waveform of each signal with the parameters T being the time vector and signal being the signal vector of chose signal (X, Y or corrected\_Y).

**Discussion of Results**

At the end of the project the original, distorted and corrected sounds are presented, listening to the .wav files of the sounds there is a clear difference between the original and distorted sounds for the distorted sound is quieter than the original and it also is nosier than the original sound. The original compared to the corrected, in the process of listening to there is no difference other than the corrected sound being a little louder than the original sound, other than that the corrected sound is free of distortions therefore validating that the inverse transfer function is a good transfer function for the use in this audio equalizing system. For bass boosting and treble boosting the sounds will be different for they will have a boost in either the lower or higher frequencies. For the bass boosted condition for all the sounds there is a clear boost to all lower frequencies which from what is heard in the song is the guitar. As for the treble boosting condition for the sounds there is a very clear increase in the higher frequencies which in the song are the bells, snare drum and the voice of the artist singing.

With the time waveforms and the .wav files there is a clear visual and audio representation of the sounds but there still needs to be a view on the power spectrum of the signals as well as the bass boosted and treble boosted versions of each. The power spectrum for the original, distorted and corrected sounds compared to the transfer function and inverse filter transfer function shows that the original power spectrum is just a given and the distorted sound is influenced by the transfer function in the way the dB/dec slope at different frequencies are shown in the distorted power spectrum and the corrected power spectrum is similar to the original for that the inverse filter transfer function is applied to the distorted sound to in simple terms cancel out the original transfer function to get the original sound that’s why there is close correlation between the original and corrected power spectrum. The bass boosted spectrums for all the sounds have a clear visual of the lower frequencies being amplified therefore having higher power for the lower frequencies than the sounds before the bass boost. For treble boosted power spectrums there is clear viewing of higher frequencies being amplified in the power spectrum as expected from the treble boosting.

**Conclusion**

In completing this project, the importance of accurate audio signal processing and the use of digital signal analysis to correct distortions in an audio system are shown to be highly enlightened. With the aid of MATLAB simulation software, the methods emphasized were possible for transfer function estimation, inverse filter design, and signal simulation. With these methods used they demonstrated their effectiveness in tackling the challenges of distortions in the output sound. The results validated the accuracy of the inverse filter, as the corrected output audio showed minimal deviations from the original input both audibly and visually, aside from minor differences in amplification over certain frequencies. The power spectrum analysis further showed the removal of distortions from the output, as the corrected signal closely matched the original signal’s power distribution across the various frequencies in these sounds. In addition, the bass boost and treble boost enhancements to these sounds effectively showcased the system's capability to manipulate specific frequency ranges, providing clear amplification on low and high-frequency components, respectively. Overall, this project demonstrated the importance of analyzing systems for deviations and correcting them if any are present. In considering distortions and implementing frequency specified enhancements, the project achieved the objective of correcting a distorted output to replicate the original as well as introducing the use of treble and bass boosting to amplify certain frequency ranges with the whole project having a strong emphasis on using MATLAB to simulate everything in the description and design of this application project.