Project Proposal
Auto Equalization

Team Members:
David Swiston
Jonathan Swiston

Project Goal:

The goal of this project will be to provide a digital based system for equalizing a pair of speakers to provide a flat in-room frequency response (for any room). This project will utilize the TMS320C54 DSP chip and will perform its function in real-time automatically flattening the in room response.

Why:

Current audio speaker technology can provide a very acceptable level of sound reproduction, often on good equipment the nearfield or anechoic frequency response of a speaker may vary by only +/- 3db variance across 20-20,000KHz. However, this close to perfect response is not what one who is listening to the speakers hear. As the sound propagates from the speaker, it not only moves directly forward towards one's ears but it spans out and meet many obstacles. The sound waves bounce off the floor, the ceiling, the walls, and any furniture in the room and also encounters resonate frequencies of the room and objects in the room. All of this interaction culminates in a farfield frequency response that can be far from the +-3db anechoic or nearfield response. As a result the sound that the listener hears is much different than intended. The sound may be bright, meaning having too much treble, or the rooms resonance frequency might be such that the bass response has a large peak. This can lead to a lackluster performance by even the most expensive and well designed speakers, unless the speakers were designed specifically for the room in which they are played. This custom design however is not practical for audio sound systems that are used in more than one location such as for concerts. It is also not economically viable for consumers who are concerned about audio quality. This makes a need for one to design an inexpensive device that can calculate the in room frequency response of a sound system and then have the device program itself to alter the input signal so as to provide an output signal that results in an in room response that more accurately resembles the input signal. This is the overall goal of our project and is why we consider it to be a valuable real-world project. It has economic interest, real world application, and is a manageable project given the time frame of the course.

Design Overview:

The next subsections provide an overview of relevant elements key to the design. They provide a description of the theory and also how we plan on implementing them.

Digital Equalization:

The system will use digital equalization by the application of digital filters. Such a system where a device automatically equalizes would be nearly impossible with analog components. However, the flexibility of digital equipment makes such a device easily realizable. We plan on using multiple high Qs filters designed in Matlab to create a "bank" of notch filters each positioned at to-be-decided octave positions and 3dB rejection bandwidths that we will pass our input through. Upon analysis of the room response, the resulting data will provide each filter in the bank with a coefficient multiplier that will provide a flat in-room frequency response. The coefficients cannot be calculated instantly by evaluating the frequency response once however. Speakers are non-linear devices so the initial filter may not produce the results predicted so after each coefficient estimate, the result will be reanalyzed [1: 2]. This will take place until a predetermined level of quality has been met.

The predesigned filters will be FIR filters so as to provide the needed performance with stability and less rounding/quantization errors. In addition, to meet the freedom necessary to test and fine tune our design, we will design our FIR filters by approximating IIR notch filters. By doing this we will be able to more easily select the 3dB rejection bandwidth and the notch frequency, and then approximate the IIR filter with an FIR filter [2: 16].

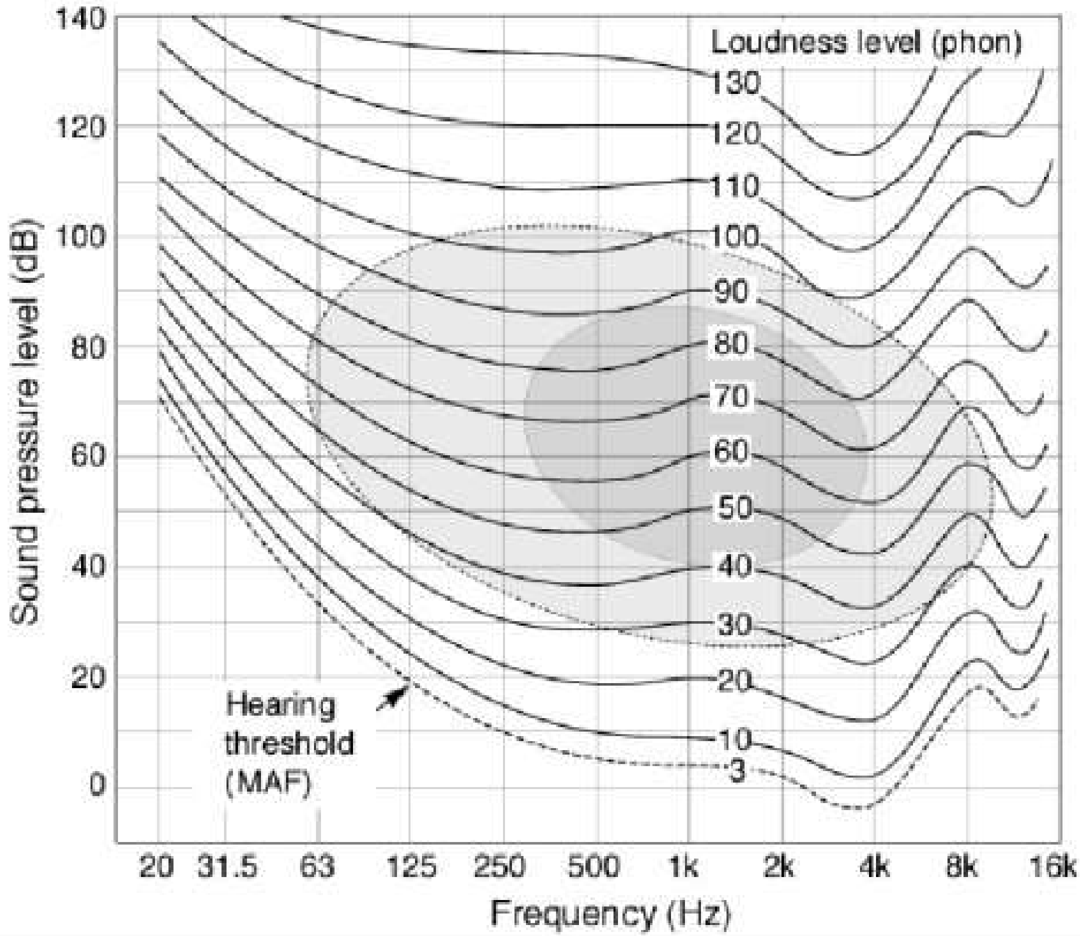
Analyzer:

In order to equalize the speakers, the power spectrum of the speaker output will be measured. Two microphones will be set up in such an orientation as to simulate the position of ears on a human head. The microphone outputs will be mixed together into one signal. This signal will be fed into the DSP where the power spectrum at specific intervals, or bands, will be computed using the FFT. Upon inspection, notch filters will be applied to the bands so that every band has an average magnitude equal to that of the other bands within a margin of error. This process will continue to loop until the desired level of quality is reached.

Pink Noise:

Pink noise will be used to equalize the speaker system. The human ear does not have flat equal loudness perception across the frequency spectrum of 20hz – 20khz.

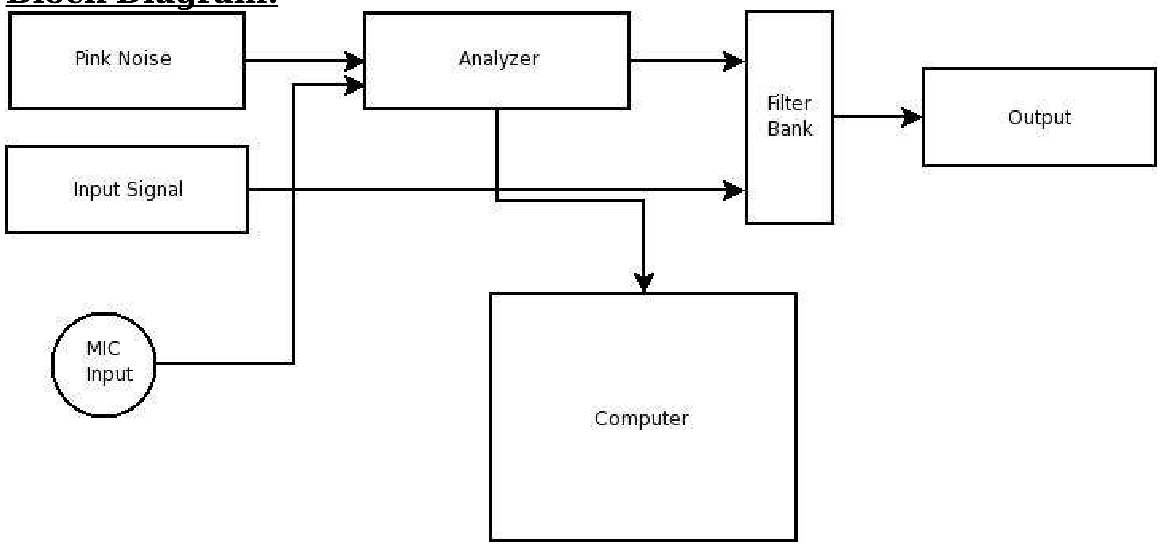
Loudness Perception of the Human Ear



http://www.acoustics.hut.fi/teaching/S-89.320/

It has been found that listeners find music produced from pink noise to be more agreable than music produced from white noise [3: 318].   The power difference between pink noise and white noise is described as follows. White noise has equal energy per frequency and pink noise has equal energy per octave. Therefore white noise has a flat frequency response while pink noise has a slope of -3dB/octave. In order to generate pink noise the Voss-McCartney algorithm will be used. In this algorithm, multiple white noise sources are produced and arranged into rows (0, 1, . . ., N-1). Each successive row is updated at half the rate as the previous row. The sum of the N rows gives the output of the pink noise generator. The equalizer will be normalized so that pink noise represents a flat frequency spectrum.

Block Diagram:



Pink Noise:

The pink noise is created by a white-noise generator and then filtered to create pink noise. It is inputted to the analyzer to provide a signal that when outputted through the filter bank, the speakers produce a flat response.

Input Signal:

The input signal is simply a source input that is the desired output. Once the analyzer is done calculating the multiplier coefficients, this signal is passed through the filter bank and then to the output so the listener is given their corrected output.

Mic Input:

This input comes from a microphone measuring the output of the speakers.   The microphone is placed in the specific spot that one is looking to equalize. This will be designated as the listening position.

Analyzer:

The analyzer uses an FFT algorithm to calculate the power spectrum of the output pink noise and attempts to level the output by changing the filter coefficients. After it makes changes it remeasures and continues to attempt to flatten the response until it meets a predetermined quality. After it has finished, it allows the input signal to be passed through the filter bank and to the output instead of the pink noise. Once it has finished calculating the coefficients, its data will be exported to the computer so that the results can be stored and in the future a simple digital equalizer can be programmed to perform the predetermined function.

Filter Bank:

This is a bank of FIR filters with predetermined notch frequencies and bandwidths which will have their coefficients multiplied by what the analyzer dictates to create a flat response. The input signals are run through this bank.

Output:

The output is simply the output of the input signal after it has been passed through the filter bank. The output is connected to an amplifier and from there, to the speakers.

Computer:

The computer provides a means to store the calculated alterations needed to be done to the input signal to provide an accurate output given the whole system response. This allows the user to replace the auto-equalization feature with a simple digital equalizer once the systems response has been calculated and equalized. It also allows one to analyze the room response and possibly make alterations to speaker placement or room layout given the response. It may also provide the user with information about room resonance.

Wrap Up:

Digital equalization is widely used. The ability to provide such a function automatically by measurement provides an economically viable product. It has a definite use and demand in the market. It is also a project that we feel can be completed in the time frame given that will allow us to reach a level of quality that will provide something meaningful. Digital noise introduced by the system is something we will be concerned about and simply getting the system functioning is not all that we are striving for. Given the large number of filters that this system is going to apply, we will also be largely concerned with the actual design and implementation of the filters and will attempt to minimize ripple, rounding, and quantization to a level that will provide high quality output.

Works Consulted:

Rusty Allred and Ryan Hsiao, "The Advantages of Digital Equalization," in International IC – China Conference Proceedings, pp. 230-237.

Suhash Chandra, Balbir Kumar, and Shail Jain, “FIR Notch Filter Design – A Review,” Electronics and Energetics, vol. 14, no. 3. p. 295-327, Dec. 2001.

Voss, R.F., and Clarke, J.: “1/f Noise in Music and Speech”. Nature 258 (1978) 317-318.