EECS 452, W.03
DSP Project Proposals: HW#5
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<u>Proposal:</u> Automated Adaptive Room/System Equalization System

Goal Product

Develop a parametric equalizer system for professional audio equipment with self-modifying parameters using a secondary microphone input. The system will modify its own parameters based on measurements it takes from the microphone input in order to time-align multiple speakers, and smooth frequency response. This would replace the need to do make the adjustments by hand. Depending on the direction of the project and difficulty the system could take measurements and make changes on the fly, or do so as a separate function using a number of methods: swept sine, pulse response, or white/pink noise.

Competing Products/Markets

A number of companies have various competing technologies that perform this process. It is aimed mostly at professional sound reinforcement applications and at “prosumer” audio professionals. Products range from $280 from Bahringer to over $1500 from Sabine. There are a number of patents that have been taken out on some devices.

Scope for EECS452

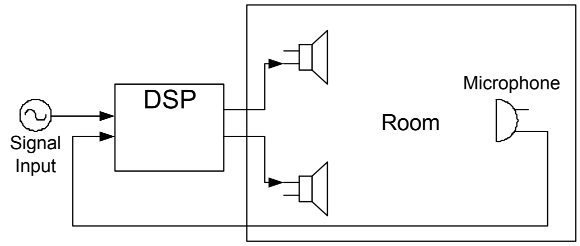
- Determine best method, or select a number of methods to determine room response and desired functions. This will involve patent searches and research.
- Use Matlab to write two programs, one to simulate a room response and one to emulate the adaptive equalizer
- Implement adaptive equalizer on DSP hardware
- Attempt to program FLASH memory
- Requires a DSP with 4x A/D channels and 2xD/A channels with at least 16-bit/44.1kHz in order to do stereo equalization and time-alignment
- Also requires speakers, a power amplifier, a high-quality microphone, and a microphone pre-amp (depending on EVM/DSK)

Deliverables

- Adaptive Equalizer implemented on DSP hardware
- Matlab simulations of room/equalizer

Discussion

There are a number of ways this project could be approached as far as equalization goes. The room measurements could be done as an individual calibration program, or they could be continuous while listening to the source material. Measurements could be based on filtered test tones, broadband noise, or impulse response, or even the source material itself. Testing could be done in various rooms around the EECS building including the RadLab's anoechoic chamber.



<u>Proposal:</u> Synthetic Thinned Aperture Acoustic Imaging (Acoustic Holography) - Proof of Concept

Goal Product

This is a proof-of-concept project that proves Synthetic Thinned Aperture Radiometry (STAR) techniques can be applied to base-band system in acoustics in order to acoustically characterize an environment (e.g. a room or theater). This involves two segments. First prove that STAR techniques on continuous single frequencies at base-band result in a higher spatial resolution than phased array techniques across a limited spectrum of audible signals (~100Hz to 20kHz). The second is to demonstrate that a multi-channel array can determine the approximate spatial location of a single sound source.

The end goal of this research is to build a multi-element (>20) STAR acoustic imager to do 2-D spatial/frequency measurements of an environment. This is a needed product and possibly a graduate level product.

Competing Products/Markets

There are very few products today that can take passive images of an acoustical environment. Some research has been put into the field of using SONAR like technologies for doing 3-d imaging, and there are a number of products to calculate room frequency and impulse response. However, there are no products on the market that I know that deal directly with source-passive acoustical holography. This is a much needed by very complex project. There is a lot of research in the acoustical holography area (Purdue), but mostly using near-field measurements and using a phased-array. Using STAR techniques would significantly reduce the hardware costs in return for increased software processing.

Scope for EECS452

- Investigate hexagonal FFT algorithms required for STAR techniques.
- Implement and demonstrate direct comparison of linear phased array vs. STAR techniques. Utilize RadLab anechoic chamber for measurements.
- Build 2-D STAR array
- Implement STAR algorithm on DSP (require many inputs, or input multiplexer)
- Write a Matlab (or other) control program for DSP control and generating output images and plots

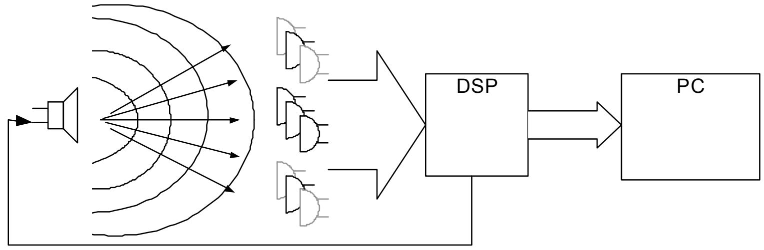
Deliverables

- Demonstrate the increased spatial resolution of STAR techniques compared to phased array (linear array)
- Demonstrate spatial quadrant source detection using 2-D array with STAR techniques.
- Propose extension on project to build a full multi-frequency 2-D STAR imager

Discussion

When setting up the acoustics and sound systems in theaters, auditoriums, recording studios, etcetera, it is important to not only have smooth frequency response, but there is a need for even spatial source strength. Also, in noise control applications it is important to locate sources of noise or locate where sound may be reflecting from. Today, sound engineers are responsible for performing the time-consuming measurements needed to measure and fix acoustic problems like these. The end result of this research and development project is to create a device that can acoustically image an environment in real-time and develop an output that can assist anyone in sound control and management applications.

This is a very ambitious project. I would love to pursue this project but it will require a lot of work. I am not sure yet exactly just how much of this can be done reasonably in EECS 452 but this would require some immediate investigation. For some information, email me, I have a number of articles and information on the subject that better describes STAR.



<u>Proposal:</u> THD Reduction in Subwoofers through Active Prewarping

Goal Product

Demonstrate an open-loop DSP based prewarping black-box that will reduce the THD levels of a standard subwoofer. Also, use the system to smooth the frequency response of the subwoofer over its pass-band. Furthermore, update the transfer function using a feedback loop indirectly using an accelerometer/position sensor on the cone of the subwoofer.

Competing Products/Markets

There are a number of servo-controlled subwoofers on the market today in the "pro-sumer" and professional marketplace. These subwoofers uses various motion controlled feedback schemes to successfully reduce distortion. However, all of these systems are very expensive, require a constant feedback element, and all rely on correcting an error after it has been produced, which is not desirable. By actively pre-warping the signal going to the subwoofer, exactly as done in newer RF communications systems, there will be no need for a direct closed-loop feedback system.

Scope for EECS452

- Through research and measurement, construct a non-linear model of a subwoofer
- Use the model to generate a method to pre-warp the subwoofer drive signal
- Use off the shelf hardware to build a test bed (Subwoofer, amplifier, accelerometer/position sensor)
- Use Matlab to simulate subwoofer and pre-warp system
- Implement pre-warp system in DSP hardware
- If time available, examine using ways of implementing automated parameter measurement
- Requires a DSP with at least 2x A/D and 1xD/A at 16-bit, >10kHz

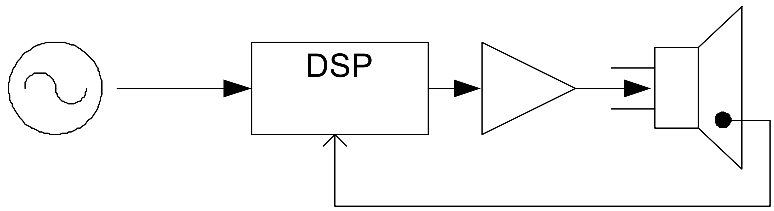
Deliverables

- Demonstrate a DSP product connected to a standard subwoofer that measurably reduces THD in the pass-band of the subwoofer
- Demonstrate automated parameter updates using secondary feedback loop.

Discussion

Subwoofers are inherently poor devices at reproducing low frequency acoustic information. The electromagnetic motor structure suffers from non-linearity due to the coil moving in and out of the static field. The cone structure also has significant mass and presents an inertia that slows the desired response. Furthermore, the cone itself can have multiple surface resonant modes that further increase distortion. Current systems aimed at reducing distortion introduce their own forms of distortion into the system due to the time delay and limited bandwidth associated with closed-loop servomechanisms.

This project is directly applicable to the current market and would be well-received, fun to do and interesting to play with. However, there is an inherent risk associated because **non-linear modeling** could be very difficult. It would be best if at least one if not more members of the project team that were involved with this had some experience or a career interest with non-linear systems.



<u>Proposal:</u> Automated THD+N Measurement System

Goal Product

Build and demonstrate a self-contained Total Harmonic Distortion plus Noise (THD+N) measurement system with capability for serial data/graph output. Include multiple modes of measuring THD including multi-frequency and averaging functions. Possibly also include frequency response measurement as well.

Competing Products/Markets

There are a number of products on the market that perform automated THD+N measurement functions either as stand-alone test-equipment or computer-based software solutions. A product to emulate would be the Keithley 2015-P, which is a stand-alone DSP based analyzer with a cost of $3495. A product description from the website is:

> … completely analyze an audio circuit in production testing. The instrument is optimized for speed and can generate frequency response and distortion [THD+N] data simultaneously. These instruments also can provide narrowband and wideband noise measurements. …

Scope for EECS452

- Determine what features are needed in a distortion analyzer, what control system is desired
- Write Matlab mockup of analyzer
- Construct interface control system (buttons, serial LCD screen, serial data interface)
- Implement analyzer on DSP hardware
- Write computer software to receive data from analyzer (Matlab?)
- Attempt to program flash memory for stand-alone operation
- DSP requirement of 1x D/A and A/D channel, higher sample rates better
- A number of electronic systems could be tested for THD measurements

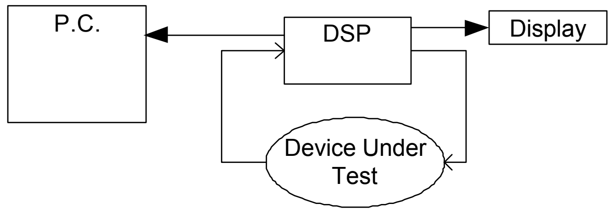
Deliverables

- Implement a DSP based THD+N measurement system

- Implement some form of computer-based software assistance for additional data output (full spectrum, full THD data)

Discussion

This project would involve low-distortion signal generation, filtering, correlation, energy summing, and external interfacing.



<u>Proposal:</u> High-Fidelity Active Parametric Mono/Stereo Crossover

Goal Product

Develop a high-fidelity 2-way parametric crossover for use in commercial and consumer sound reproduction equipment. A crossover sends individual frequency bands (usually low and high) to the respective speakers (usually subwoofer and tweeter). The crossover will be time aligned and have parametrically variable filters. The end result also should be scalable, to include more than just two frequency bands. In order to be truly a useful product, it would be beneficial to use the on-board flash memory to be truly stand-alone

Competing Products/Markets

There are dozens of companies that make active crossovers (both analog and digital) for use in consumer and professional audio equipment for use with multi-amplification systems. This product would be in direct competition with those products but possibly be cheaper and smaller. In the end, something like this product might be integrated into a home A/V receiver or other consumer electronic device (unlike a stand-alone crossover)

Scope for EECS452

- Develop requirements and specifications for crossover
- Determine best way to do filtering/time-alignment
- Simulate DSP functions in Matlab
- Implement one or two channel crossover on DSP hardware
- Attempt stand-alone operation using FLASH memory

Deliverables

- Working mono (possibly stereo) parametric 2-way crossover

Discussion