Reduction of Low Frequency Loudspeaker Distortion Using an Adaptive, Driver Independent Digital Signal Processing System

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The Loudspeaker
Acts as a piston. Pushes air forward and backward to create the longitudinal waves which produce sound.

Spider
Provides the main restoring force for the cone. Must be stiff enough to resist reverberation, but pliable enough to allow the cone to move.

Surround
Prevents the cone from wobbling. Must be stiff enough to stabilize the cone, but pliable enough to allow the cone to move forward without buckling.
Sources of Distortion

Non-linearities of Driver Materials
The materials which make up the driver are inherently non-linear. This contributes to harmonic distortion. The spider and surround must be chosen carefully since they are integral in supporting the cone.

Fringing Of the Magnetic Field
The magnetic field between magnet and pole piece is not completely contained within the gap. Fringing occurs above and below the gap leaving asymmetrical forces to act on the voice coil.
Analog Distortion Correction

Although these systems work they are highly driver specific. The transfer function of the driver must be estimated in order to calculate the transfer function H(s), which is then hard wired as an RLC circuit.

Drawbacks

Analog Distortion Correction

Similar to a system implemented by Velodyne. Standard analog control system using an accelerometer to feed back information about the true motion of the cone, and a corrective transfer function in the feedback loop to reduce distortion.
MCM Audio Select 15" Sub-Woofer

- $f_s$: 24Hz
- SPL: 93dB
- Voice Coil: 3"
- Peak Power: 500W
- RMS Power: 250W
- Impedence: 8Ω

Frequency Response of Driver
Digital implementation of a filter. The signal is processed by going through a series of multiply and accumulate operations.
Adaptive Control Algorithm

Adaptive Filter

Tap weights are variable

This filter changes along with the data. The new tap weights are calculated according to the output of the filter and the desired signal.
Based on feedback - produces an error signal which is the difference between the output of the filter and the desired signal. This error signal is then used to calculate the new set of tap weights.

Define:

\[ \mathbf{w}(n) = [w_0 \ w_1 \ w_2 \ w_3 \ w_4] \]

\[ y(n) = \mathbf{w}^H(n)\mathbf{u}(n) \]

\[ e(n) = d(n) - y(n) \]

\[ \mathbf{w}(n+1) = \mathbf{w}(n) + \mu \mathbf{u}(n)e^*(n) \]

**LMS Algorithm**

- **w(n)** = Tap Weight Vector
- **u(n)** = Filter Input
- **y(n)** = Filter Output
- **e(n)** = Error Signal
- **d(n)** = Desired Signal
- \( \mu \) = Step Size Parameter
Proposed system similar in implementation to an analog system, however the filter controlling feedback is adaptive, changing along with the signal output from the loudspeaker and therefore independent of any particular driver.
Accelerometer

Compression Mode Accelerometer

In this accelerometer the quartz crystals are sandwiched between a post a seismic mass. When a compressive force is exerted on the crystals by the mass the crystals produce a charge.

Shear Mode Accelerometer

In this accelerometer the quartz crystals are sandwiched between a post a seismic mass. When a shear force is exerted on the crystals by the mass the crystals produce a charge.

Quartz Crystal

The quartz crystal is heart of the accelerometer. Different types of cuts are made from the crystal to produce accelerometers which operate in different modes.
Additional Equipment

Accelerometer Amplifier

The output of the accelerometer is in the millivolt range. This output had to be amplified so that the two signals being compared by the adaptive filter are of approximately equal magnitude.

Specifications

Passive High Pass Filter:
\[ f_{3dB} = 3\text{dB} \]

Active Variable Amplifier:
Gain: .1dB - 20dB
The output of the adaptive filter had to be low pass filtered before being played on the loudspeaker. This filter was implemented to complete this task.

Specifications

Active Low Pass Filter:
- $f_{3dB} = 99\text{Hz}$
- Gain = 3.5dB
DSP System

Spectrum PC/C5x Board
Contains a Texas Instruments DSP320TMSC50 fixed point digital signal processing chip operating at 40MHz. A 16-bit processor which preforms 20 million instructions per second.

Spectrum AM/D16SA ADC/DAC Board
 Implies 16-bit conversion of two separate channels at a sampling rate of up to 200kHz using the succesive approximation method, while providing very low addtional distortion (SNR = 110dB).
Before actually implementing the adaptive filter on the DSP board a simple assembly program was written to test the board's I/O capabilities. In this case the input was a simple sine wave and the output is a sampled version of that wave.

This was an important test to run as it eliminated the possibility that the filter was not working because the board did not work.
Laboratory Setup
Before the LMS algorithm converges the output of the filter is different from the input.

After the LMS algorithm converges the output of the filter closely matches the input.
Simulation

The input is modeled as a signal containing several frequency components. The output of the loudspeaker is modeled as this signal with additional frequency components due to the distortion introduced by the loudspeaker. THD = 34.28%

Spectrum - Signal & Distortion

The input is modeled as a signal containing several frequency components. The output of the loudspeaker is modeled as this signal with additional frequency components due to the distortion introduced by the loudspeaker. THD = 0.14%

Spectrum - Signal & Filter Output

The input is modeled as a signal containing several frequency components as above. The output of the filter is computed in MATLAB and shows a significant reduction in distortion. THD = 0.14%
Measurements

Signal generator produces a very clean signal. Second harmonic component of a 15Hz sine wave is 70dB down. THD = 0.00001%

Power Spectral Density of Signal Generator

Output of loudspeaker is very distorted, with 1\textsuperscript{st} through 5\textsuperscript{th} harmonic components at unacceptable levels. THD = 23.87%

Power Spectral Density of Loudspeaker
Final System Configuration

After countless tries at countless configurations the final configuration decided on was the above. This system implements two adaptive filters. The second prefilters the signal and the first takes the filtered signal and the output of the accelerometer and uses them to compute the new set of tap weights. The second filter is then updated with the same tap weights.
Assembly Program

Set Up Sampling Rate

Set Up Interrupts

Set Up Circular Buffers

Wait for Interrupt

Read New Samples (CD and Accel)

Put Data in Buffers

Filter CD, Send to Speaker

Filter Speaker Output

Compare Output and Input, Produce Error Signal

Compute New Tap Weights

Return to Wait State
Results (15Hz Sine Input)

- **Input**
  - Power Spectral Density - Unfiltered Signal

- **Output**
  - Power Spectral Density - Filtered Signal
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