Project Title: DSP Implementation of a 1961 Fender Champ Amplifier
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Advisor: Dr. Stewart
Memo: Functional Description

Introduction

This project will use Texas Instruments' TMS320C6711 32-bit floating-point DSP to generate the transfer and distortion characteristics of a 1961 Fender Champ Amplifier at its 12 different volume settings. Audio signals either saved to file or generated by a guitar via an A/D interface to the DSP will be passed through C/C++ digital filters with different gains. Unique gains for each filter in addition to the averaged nonlinear transfer characteristic for each filter's frequency range will reproduce the frequency response of the tube amplifier. Nonlinearities produced from this configuration are thought to be the primary reason for the improved quality of sound. The intention is to produce a vintage tube amplifier's sound with a low-cost DSP as opposed to paying $1000 and up for a rare amplifier with limited replacement components.

Inputs/Outputs

The system inputs will be an analog audio signal from either a guitar A/D interface or a saved audio file, and software based volume selection will regulate the filters' behavior. The output will be an audio signal with similar distortion as a tube amplifier.

Modes

The system modes will consist of the 12 volume settings similar to those provided with the 12-volume switch on the 1961 Fender Champ.

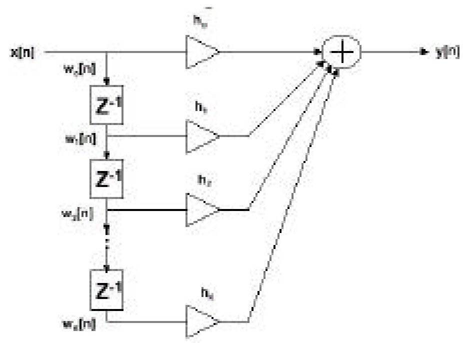
Methods

Two methods will be used to obtain the sound of a 1961 Fender Champ:

- One approach will consist of eight digital filters each cascaded with their own nonlinear transfer characteristic. The output of each nonlinear model will be filtered again to reduce any high frequency components that are not present from the 1961 Fender Champ data. The eight filtered results will be summed together and filtered once more by a bandpass filter with a wide passband to reduce any further high frequency terms from the summation operation and any DC offset. Each input sample will be placed in memory and processed on a sample-by-sample basis as each input becomes available. The result will be a real-time filter with infinite duration once the DSP is initialized.

- Another possible approach to the digital filtering algorithm is multirate signal processing where the frequency response is broken into several subbands using a multistage lowpass and highpass filtering network. Since the essential characteristic that determines the tube amplifier's sound is thought to be its nonlinearities, this method can provide the means for applying distinct nonlinear transfer characteristics of the Champ in limited frequency ranges in order to effectively model the amplifier's nonlinear network. However, there is an inherent delay from the cascading FIR filter structure that may dictate the use of the first method.

The digital filter bank will be implemented with a 2nd or higher order FIR filter design. If the algorithm's noise floor is greater than the 16-bit input's noise floor, the FIR model will provide the best response. The figures on the next page show the basic FIR filtering algorithm model, and the system block diagram. Finally, all executable code for Texas Instruments eXpressDSP™ software will be developed with MATLAB 6.5's Embedded Target for the TI C6000 DSP feature in Simulink®. References are listed on the following page for verification of the third design approach, of public and professional interest, and of continued investigation to the present moment.



$$y[n] = \sum_{k=0}^{\infty} h[k]x[n-k]$$

Figure 1: FIR Digital Filter Algorithm

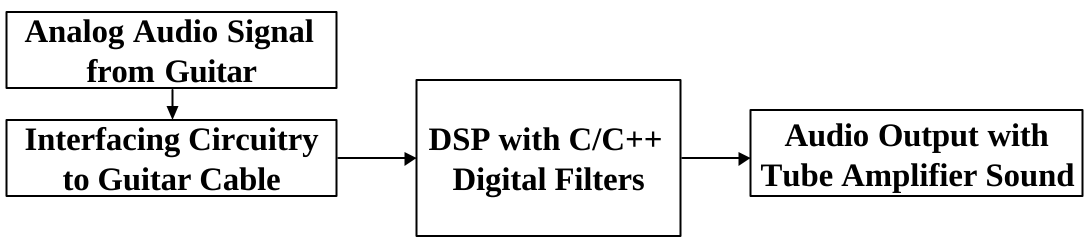


Figure 2: System Block Diagram

References

Digital Signal Processing: Principles, Algorithms, and Applications. John G. Proakis, Dimitris G. Manolakis. Third Edition. Upper Saddle River, New Jersey: Prentice Hall, 1996 pp. 832-834.

Barbour, Eric. "The Cool Sound of Tubes." Ed., Michael J. Riezenman. IEEE Spectrum August 1998. 1998. Google. IEEE. 12 pp. Google. 11 Oct 2002. <http://www.spectrum.ieee.org/select/0898/tube.html>.