Project Title: DSP Implementation of a 1961 Fender Champ Amplifier
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Memo: Complete System Level Block Diagram

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. Two approaches will be used to reproduce the 1961 Fender Champ's unique sound. Software will enable one of the 12 amplifier volume settings.

In the first method, eight digital filters will be 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 remove 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.

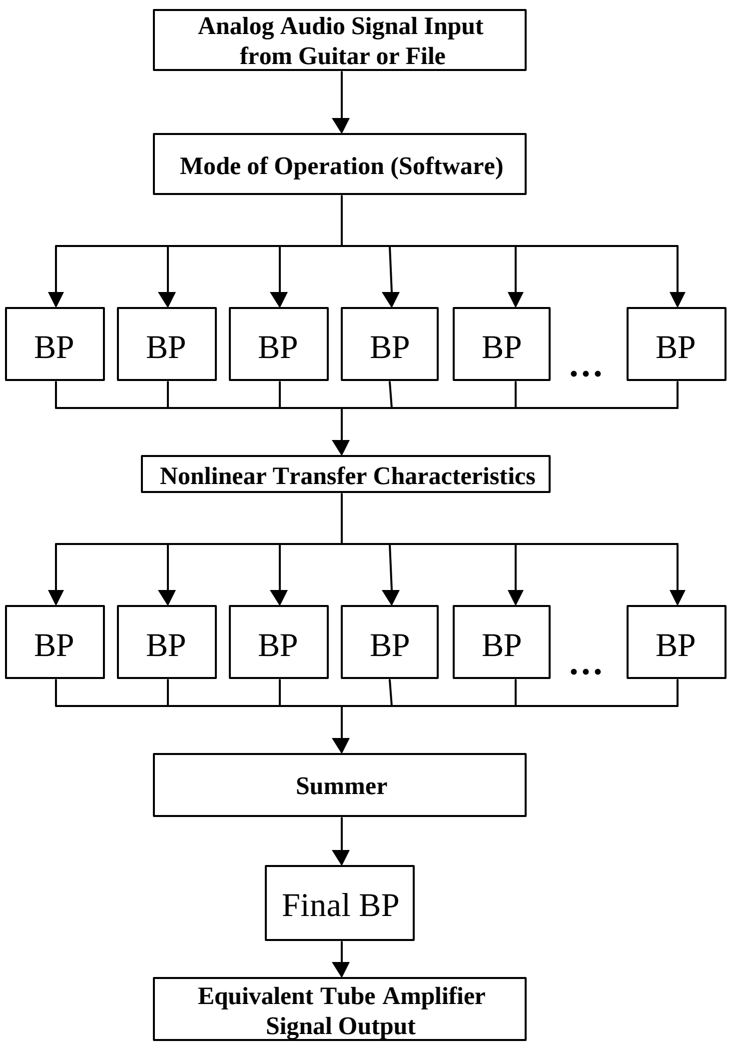

Figure 1: Parallel FIR Digital Filter Approach to Tube Amplifier DSP Model

The final and most likely approach is multirate signal processing [1]. In this scheme, the frequency response of the input is broken into several subbands with a multistage lowpass (LP) and highpass (HP) FIR filtering network. This first stage is defined as the Analysis Filter Bank. Nonlinear transfer characteristics, distinct for a range of frequencies, are then applied to each of the Analysis Filter Bank's subbands. The final output is then reconstructed by decoding the encoded frequency bands, up-sampling the response, and summing both FIR filtered results to be up sampled by another stage. This final network is the Synthesis Filter Bank. Thus, this method can provide the means for modeling the Champ's nonlinear network while processing a wide range of frequencies from the guitar input with lower order FIR filter designs. However, there is an inherent delay from the cascaded FIR filter structure that may dictate the use of the first method. This approach can be seen in the figure below.

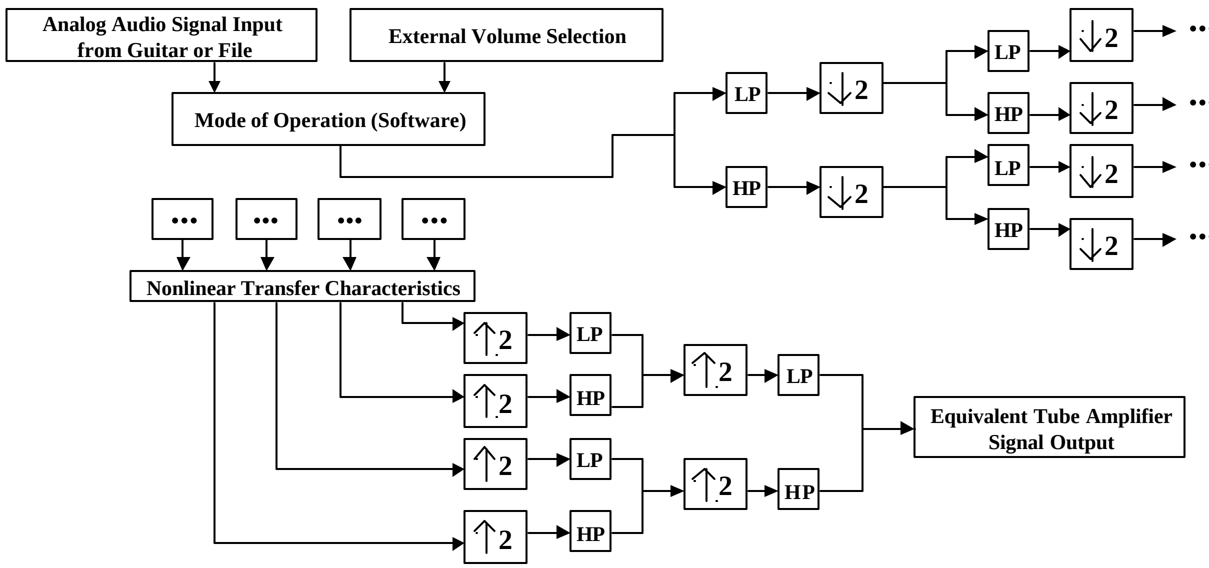

Figure 2: Multirate Signal Processing Approach to Tube Amplifier DSP Model

References

[1] 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.

[2] 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>.