Digital Modeling/Implementation of Valve Amplifiers

Senior Project Final Report

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I. Introduction

It is estimated that seventy five percent of the market for vacuum tubes is taken up by electric guitar amplifiers. Since tubes were first used in guitar amplifiers in the 50's and 60's, their signature sound has become a staple for rock guitar players. Valve amplifiers come in a broad variety of shapes and sizes. Every model has its own signature tone that musicians specifically seek out. When it plays loudly, the sound of vacuum tube amplifier arguably presents much better than the sound from a solid state amplifier because of the more musical and smoother clipping. Also, since there are much greater operating voltages for valves, they have a much greater dynamic range. Unfortunately vacuum tubes are large components that overheat often. In addition, tubes have a short life span and they can burn out at any time. Sometimes this even can cause damage to the transformer and other components. Once they burn out, they often need to be replaced quickly. This is caused by the higher power consumption that results in a lot of heat being generated within the valves. The downside of using a tube amplifier is not only the overheating and aging problems. These pieces of equipment can be too heavy to be moved with ease. This problem is compounded by the fact that most bands have multiple guitar and bass players. Most of them need their own amplifier. The cost of transporting and maintaining these tube amplifiers can be a burden on the bands that use them. Musicians are constantly in need of gear that not only is lightweight and cost effective, but gives them the ability to have any tube amplifier at their disposal for the wide range of gigs and sessions.

II. Motivation and Objectives

Due to the inconvenient nature of the valves themselves and the need to preserve their highly sought after tone, digital implementation is a viable alternative. Affordable, convenient, and versatile equipment is what many musicians are looking for in their gear. Modeling vacuum tubes and their wide dynamic range digitally can eliminate the problems that valve amplifier owners face while maintaining all of the subtleties that valve amplifiers offer. This technology gives the musician the option of using multiple amplifiers. Digital implementation will always be more cost effective and versatile than previous years. Not only will the musician have the option to incorporate more amplifiers into their arsenal, they will also have the benefits of lightweight equipment that doesn't overheat.

III. Methods

3.1 Black Box

Originally, a black box approach was considered for modeling the tube stage of the guitar amplifier. The benefit of a black box approach is that the number of components in the circuit is irrelevant, all that matters is the mapping between the input and output. Figure 1 shows the black box system diagram.

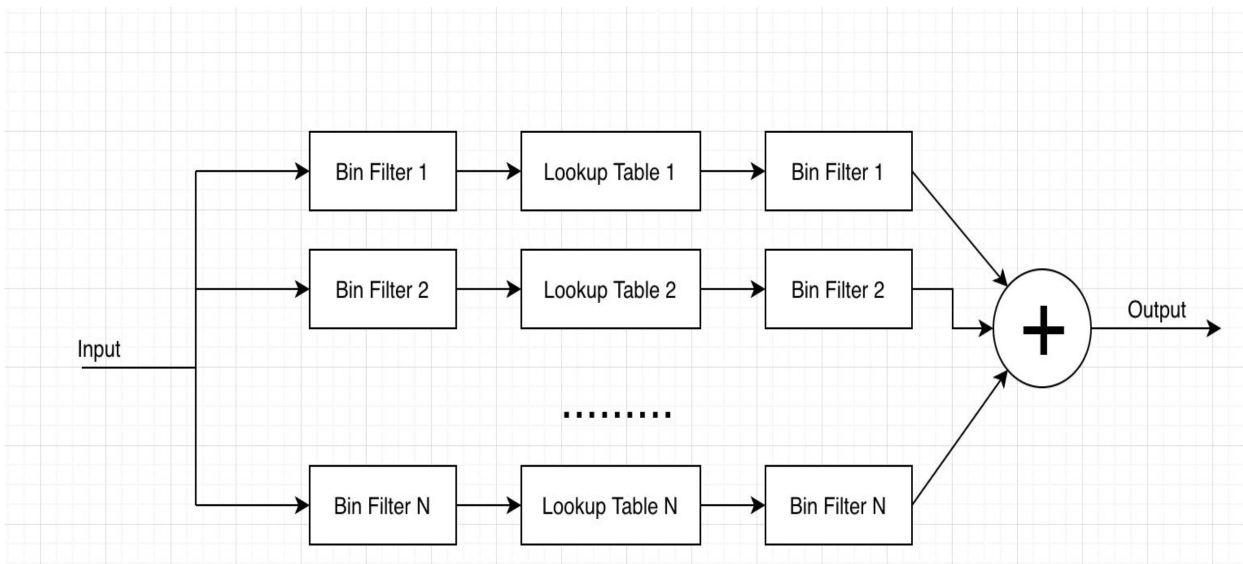

Figure 1. Frequency Bin System Block Diagram

As seen in Figure 1, the input is fed into a set of bandpass filters called bins. Each bin then is processed using a lookup table based on the bins characteristics. The result is then fed back into an identical bandpass filter removing unwanted distortion. Then the results from each bin is summed together to produce the overall system output. The problem with this method is that it does not take memory into account. The system works for if the system is static, but if we look at Figure 2 below, a typical triode stage amplifier contains capacitors in the circuit. This means that the mapping between between the input and output is not one to one, making the black box approach not viable.

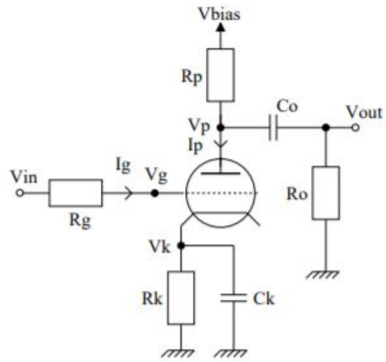

Figure 2. Triode Stage Amplifier Circuit

3.2 Wave Digital Filtering (WDF) Method

Wave Digital Filtering is a way of modeling physical systems or analog circuits in such a way that filter coefficients or components can be easily adjusted in order to change the behavior of the model. As opposed to Kirchhoff circuits whose parameters are current and voltage, Wave Digital Elements have parameters called wave quantities. There are many different sub classes of Wave Digital Filters that are used to describe different models. The different components are connected using ports. The mapping of the individual components is shown in Figure 3 as referenced in [1]. Serial and parallel adapters are used to place components accordingly. Wave Digital Filters are modular, so they can easily be altered which is good for implementation of valve amplifiers. Theses filters also have a very good dynamic range performance and have guaranteed stability under linear conditions. The nonlinear portion of the valve amplifier will be modeled through look up tables in order to match the behavior of the original model.

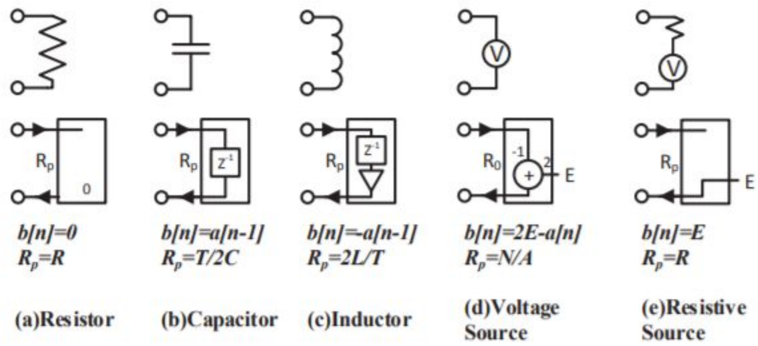

Figure 3. WDF Elements

The signal flow diagram for WDF adapters are also shown in Figure 3 as described in [4]. The final product uses a combination of multi-port serial and parallel adapters. Two port adapters will be implemented similarly. The symbol for series and parallel adapters can be shown in Figure 4. The system is then be compared with the original digital equivalent. Modularity of the structure as well as the ability to change the filter coefficients are both important criterion as well for modeling any given tube amplifier. Simulation results have been found for the triode valve's nonlinear parameters using the Leach model as well as the Norman Koren Model.

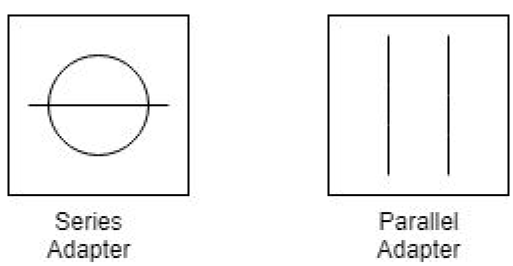

Figure 4. WDF Adapter Symbol

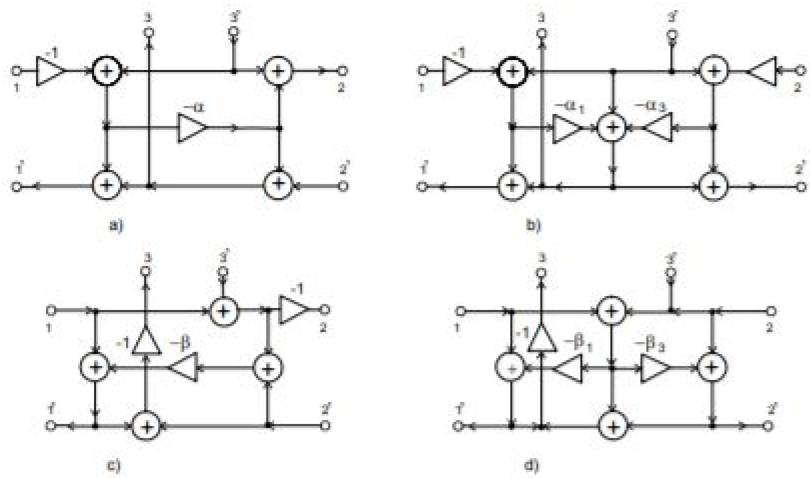

Figure 5
WDF adaptor Signal Flow Diagram: a) parallel reflection free adaptor, b) parallel adaptor, c) series reflection free adaptor, d) series adaptor.

IV. System Block Diagram

The system block diagram is shown below in Figure 6. The wave starts at the components of the system. The wave is then propagated up to the root. In our case the Nonlinear Triode is chosen as the root. Once the wave reaches the root, the numerical analysis is performed to replicate the nonlinearities of the Triode valve. After this, the wave propagated back down the system. Then all outputs are gathered as the process repeats. The series and parallel adapters have been chosen according to the triode amplifier design. This can be visualized by noting what fashion (series or parallel) the resistors and capacitors are connected on the initial analog schematic. For example, the cathode circuit has a resistor and capacitor connected in parallel. In the WDF equivalent system schematic, these two components are clearly connected to a parallel adapter.

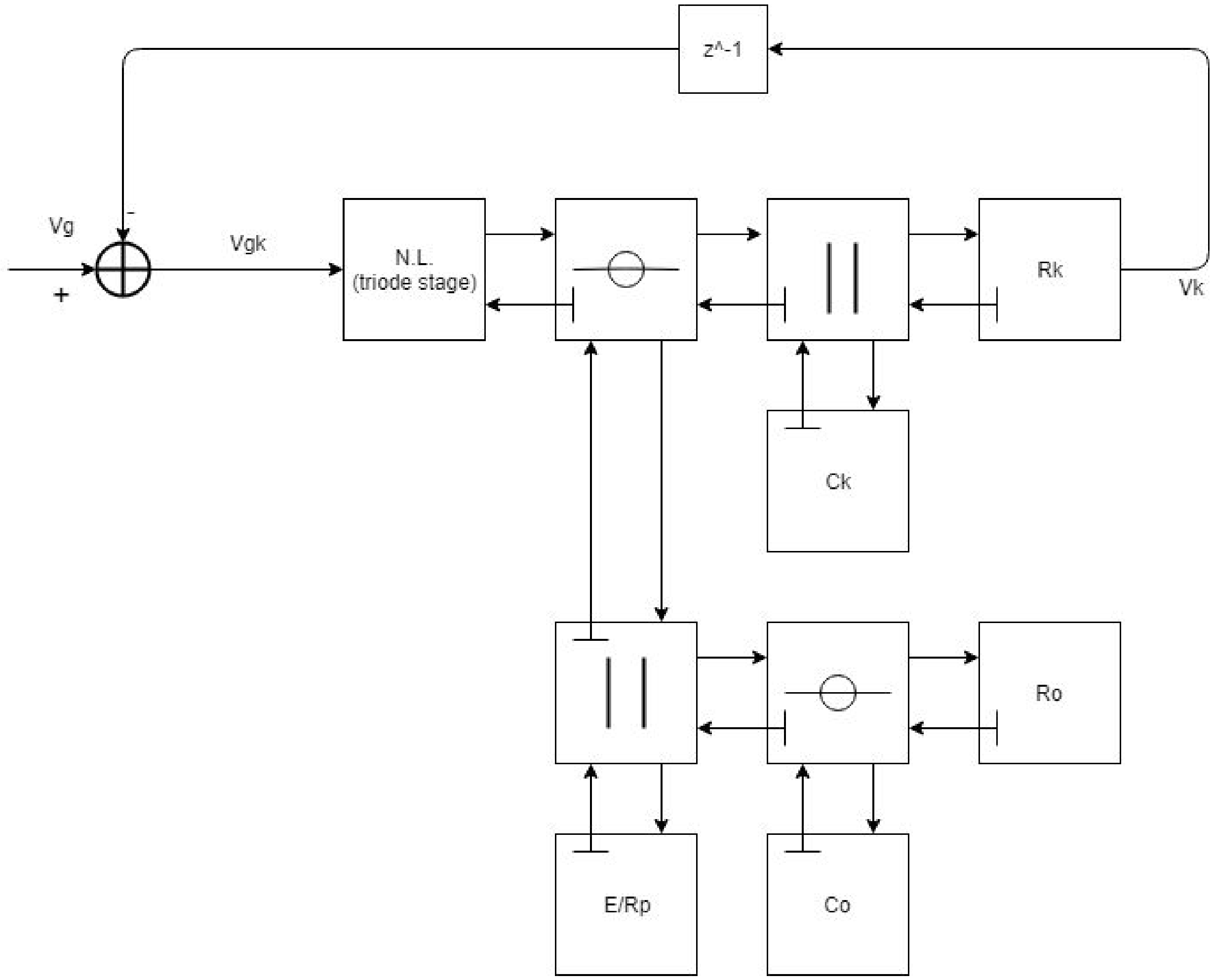

Figure 6. System Block Diagram

V. Experimental Results

Tube characteristics are modeled according to Leach's model as well as Norman Koren's model. Leach's model gives the current as a function of Vgk (voltage from grid to cathode) and Vpk (voltage from plate to cathode) for a common cathode triode tube. The equations for the leach model are described in equations 1 and 2 respectively. The simulation results for the Leach model is shown by Figures 7 and 8. For the Norman Koren model, a different set of equations are used. They are referred to as equations 3 and 4 respectively. The Norman Koren model is derived from the Leach model. The constants in these equations as well as the equations themselves can be found in [5]. Simulation results for the Norman Koren model can be found in Figures 9 and 10.

$$
\begin{aligned}
L_p &= \begin{cases} K(\mu V_{gk} + V_{pk})^{3/2} & \text{if } (\mu V_{gk} + V_{pk}) > 0 \\ 0 & \text{else} \end{cases} \\
L_g &= \begin{cases} 0 & \text{if } V_{gk} < V_\gamma \\ \frac{V_{gk} - V\gamma}{R_{gk}} & \text{else} \end{cases}
\end{aligned}
$$

Equations 1, 2

$$
\begin{aligned}
E_1 = &\quad \frac{V_{pk}}{K_p} \ln\left[1 + \exp\left(K_p\left(\frac{1}{\mu} + \frac{V_{gk}}{\sqrt{K_{vb} + V_{pk}^2}}\right)\right)\right] \\
L_p = &\quad \frac{E_1^{E_x}}{K_g}(1 + \operatorname{sgn}(E_1))
\end{aligned}
$$

Equations 3, 4

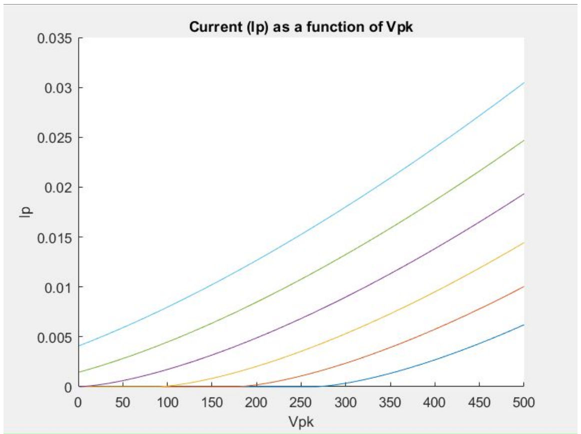

Figure 7. Plate current as a function of Vpk (Leach's model)

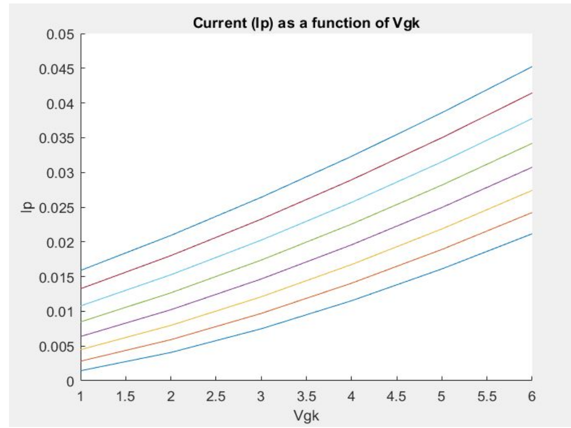

Figure 8. Plate current as a function of Vgk (Leach's model)

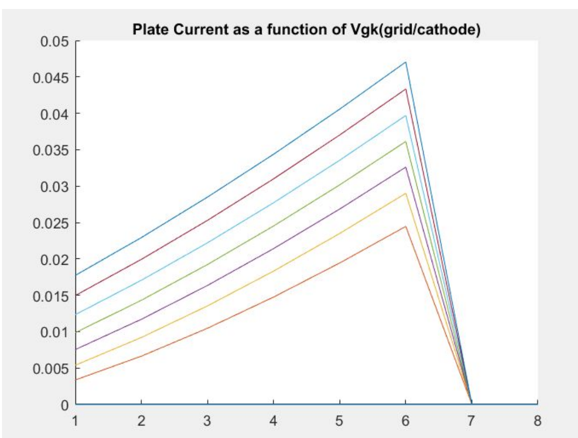

Figure 9. Plate Current as function of Vgk (Norman Koren model)

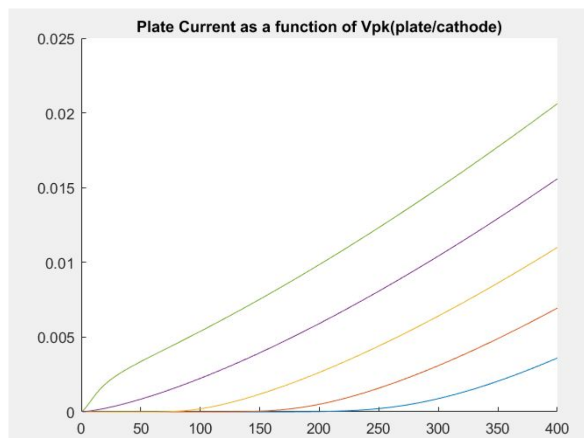

Figure 10. Plate Current as function of Vpk (Norman Koren model)

In order to verify that Wave Digital Filtering is a variable method, a simple low pass filter was constructed. The WDF equivalent is shown in Figure 11.

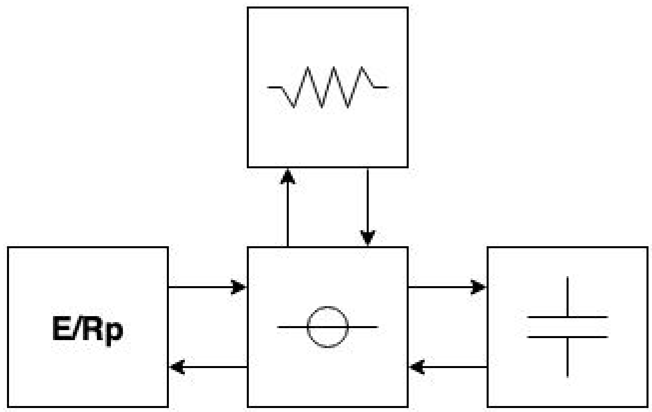

Figure 11. Simple WDF Low Pass Filter

The Filter consists of a resistive voltage source, a capacitor, and a resistor connected in series. The following components are set accordingly: **E**=1Vpp, **Rp**=1Ω, **C**=35uF, **R**=10Ω. The -3dB corner frequency is around 454.7Hz. To verify the WDF model, a Matlab script was generated to simulate the results. As shown in Figure 12, the corner frequency for the Matlab model is around 422 Hz. This is due to the resistive voltage source. Rp and R connect in series with a resistance of 11Ω. The corner frequency is set around 422 Hz. The Matlab results were cross checked with a spice simulation to further verify the results. As shown in figure 12, the corner frequency is around 422 Hz. This is due to the resistive voltage source. Rp and R connect in series with a resistance of 11Ω. When changing to 11Ω in LTspice, the corner frequency is set around 422 Hz.

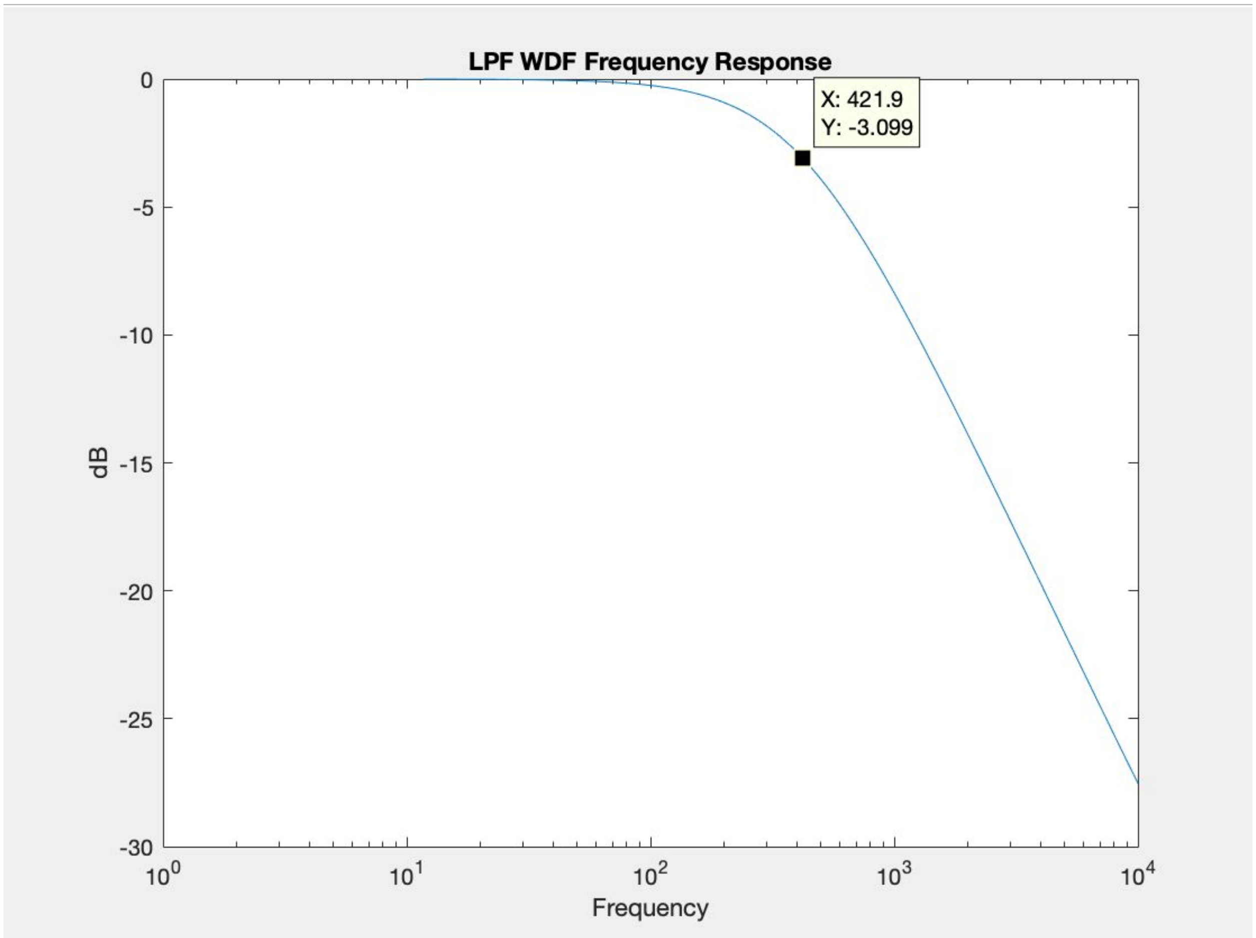

Figure 12. WDF LPF Results

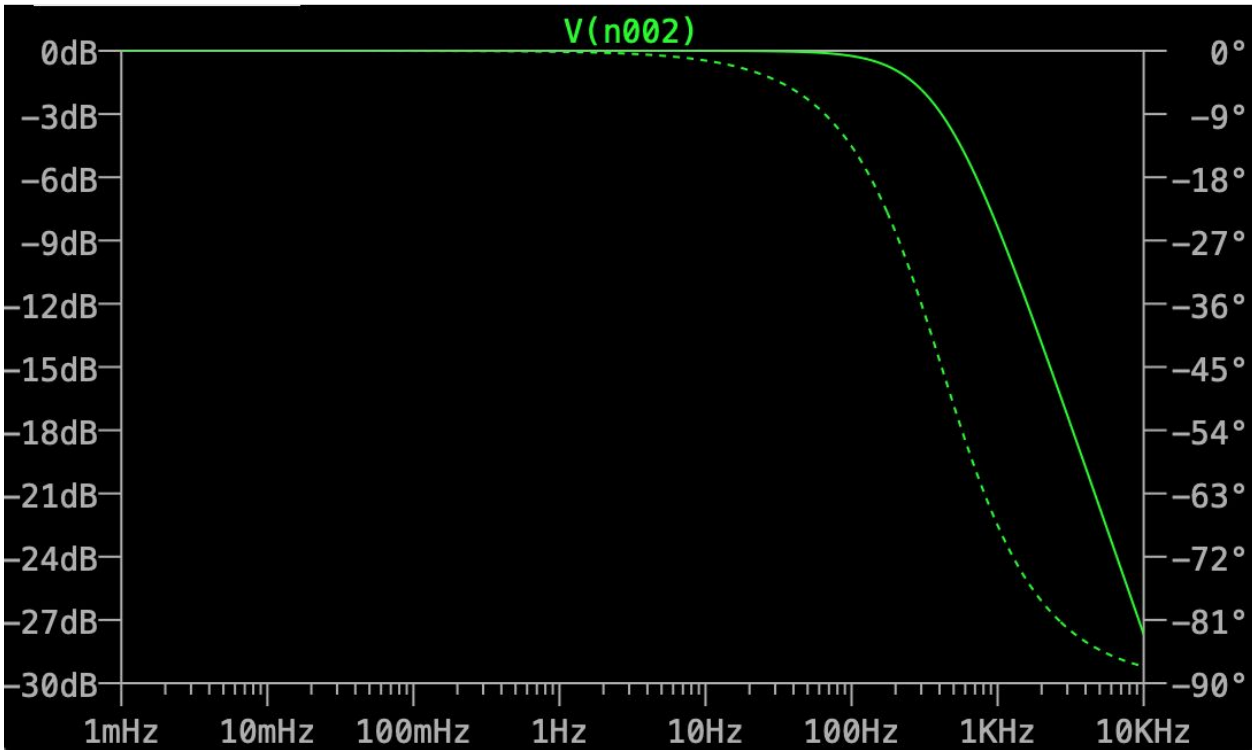

Figure 13. LTspice LPF filter

By comparing Figure 12 with Figure 13, it can be concluded that Wave Digital Filtering is a viable method for circuit model. A script for the overall system (refer to Figure 4) was generated using Matlab. The system was tested by exciting the system with a sine wave at the input terminal. Results are shown in Figure 14. To validate the results, the WDF circuit in system circuit is constructed in LTspice. Results from LTspice are shown in Figure 15. When comparing Figure 14 and Figure 15, the results are identical.

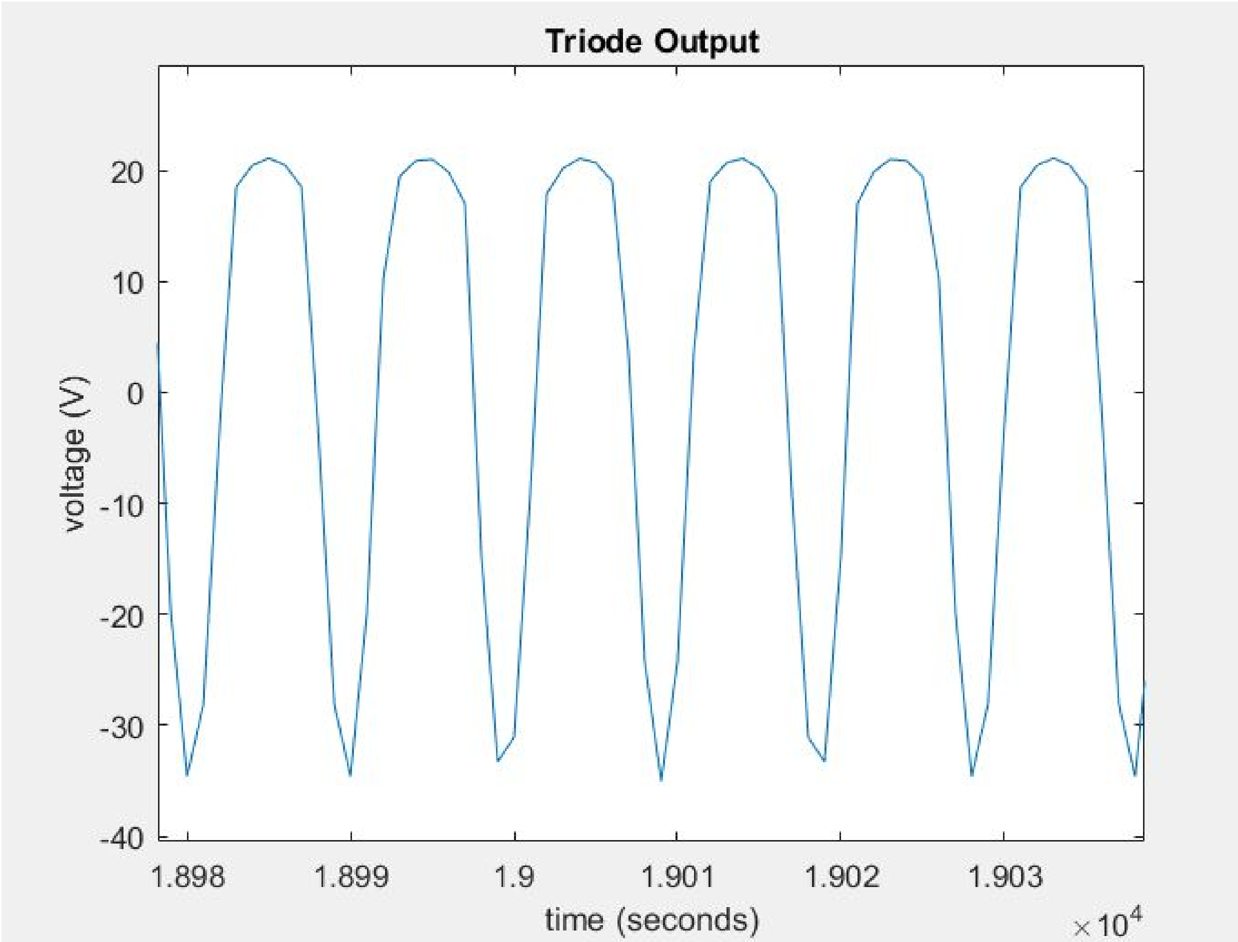

Figure 14. Triode Stage WDF circuit output

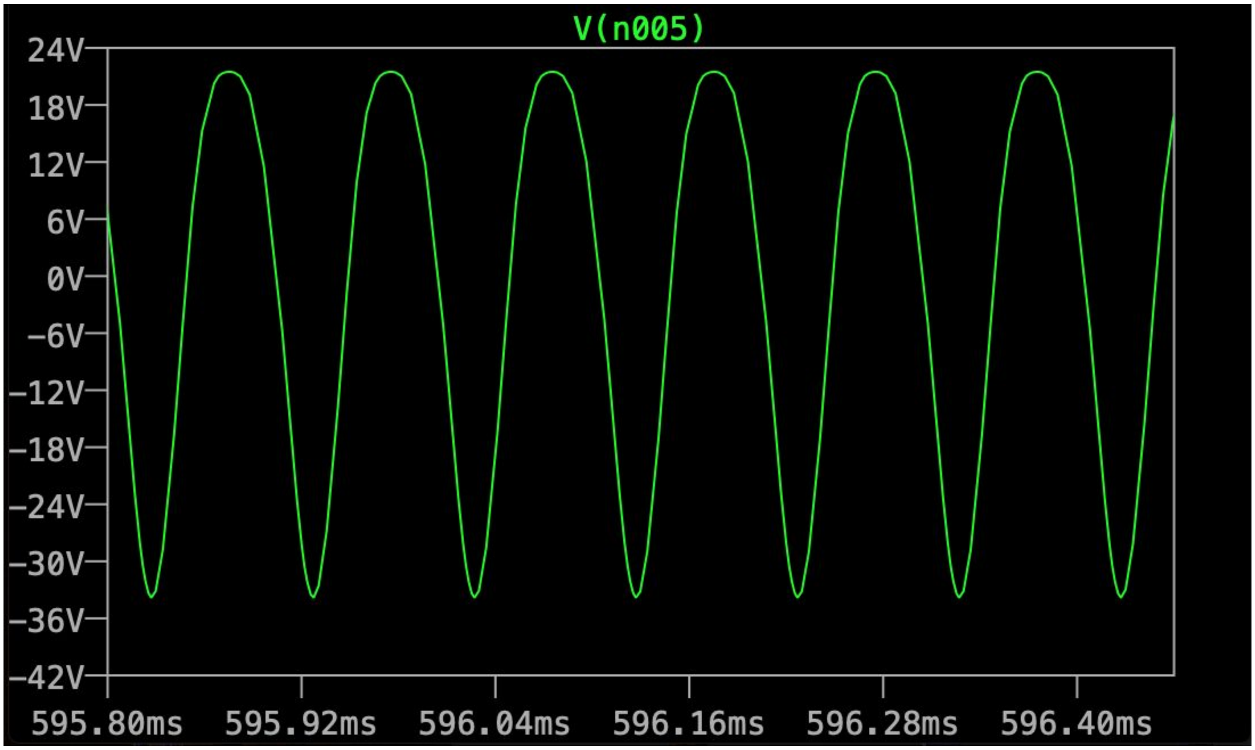

Figure 15. Triode Stage LTspice circuit output

With the circuit validated, the system can now be implemented in real-time. Instead of a sine wave, the input is generated with an actual electric guitar. To obtain the guitar samples, an audio interface was used. To simplify the process, The audio systems toolbox was used to read the data from the audio interface. Since the system memory requires a maximum delay of 1 sample, implementing the system in real-time is straight forward. An infinite while loop performs the following, read audio interface data, perform WDF on sample frame, and write data to audio interface. Figure 16 represents the output of the system excited with a G chord performed on an electric guitar (orange). The figure also compares the results with an LTspice simulation (blue). The figure shows that the WDF system almost maps the signal one to one with the LTspice simulation. The slight variation in both signals is due to the different numerical methods between the two programs.

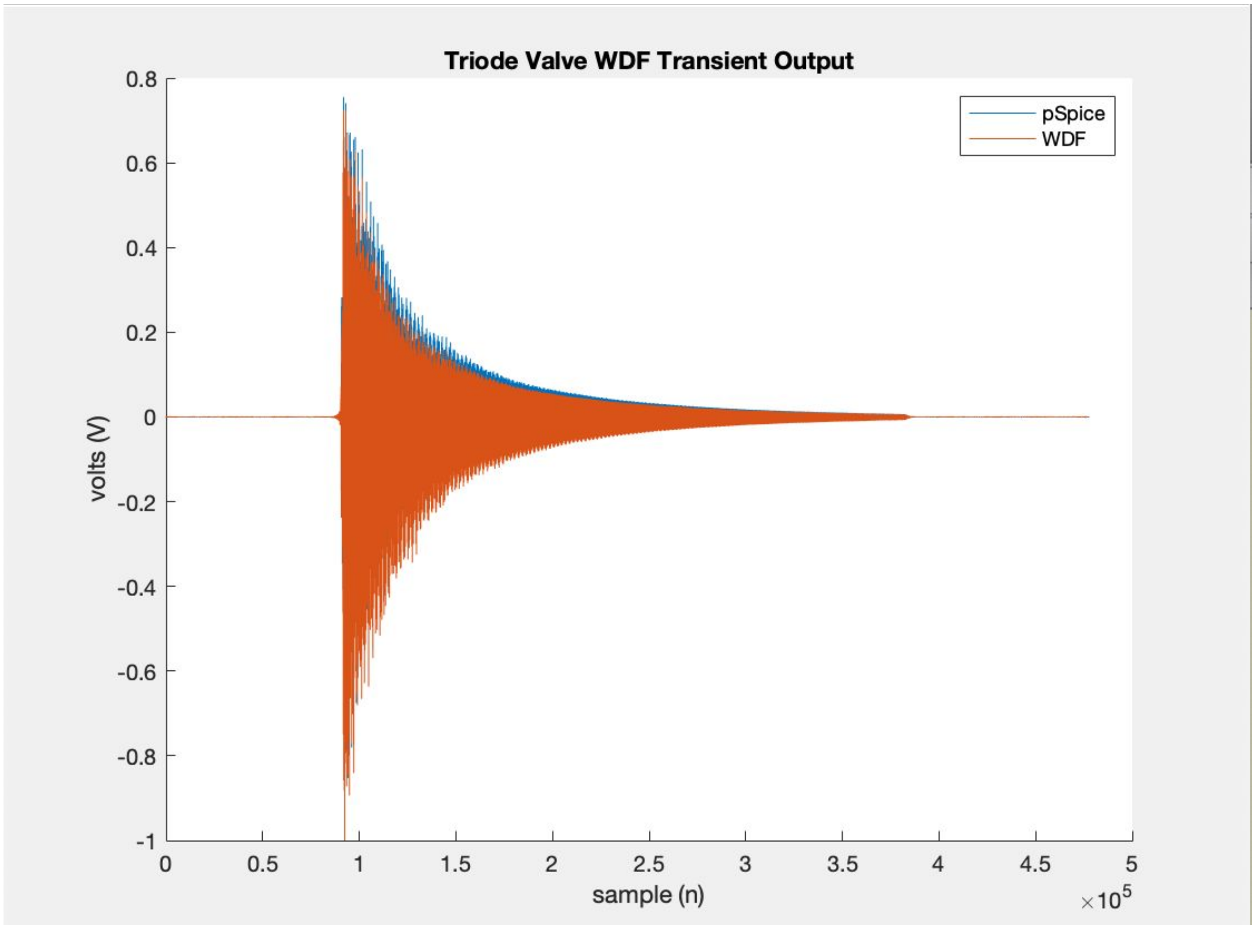

Figure 16. WDF Guitar signal

VI. Conclusion

This project designed a digital emulation of an analog valve amplifier with the non-linear characteristics being reconfigurable. Emulating the overall response of the valve amplifier is performed using the Wave Digital Filtering technique. A single stage Triode Amplifier can be modeled using WDF as shown in figure 4. The Triode component itself is calculated using a Newton-Raphson solver method. The Triode current is numerically modeled using either the leach or Norman Koren models. After simulating the system through sine wave and guitar excitation, we can conclude that a single stage Triode amplifier circuit and be accurately emulated using Wave Digital Filters. As for future work. Emulating cascaded triode stages can be performed using a new WDF adaptor called the Rigid adaptor. Not all circuits can be broken down into series and parallel connections. The Rigid adaptor is an adaptor that is designed to emulate all other connections that cannot be categorized as series or parallel.

VII. References

[1] M. Karjalainen and J. Pakarinen, "Wave Digital Simulation of a Vacuum-Tube Amplifier," *2006 IEEE International Conference on Acoustics Speech and Signal Processing Proceedings*, Toulouse, 2006, pp. V-V. doi: 10.1109/ICASSP.2006.1661235

[2] S. D Angelo, J. Pakarinen, and V. Välimäki. New Family of Wave-Digital Triode Models. IEEE Trans. Audio, Speech, and Lang. Process., vol. 21, no. 2, pp. 313 321, February 2013.

[3] Bohumil Psenicka, Francisco Garciá Ugalde, and Andrés Romero, “Design of Wave Digital Filters,” *Universidad Nacional Autónoma de México*, December 2009

[4] M. Antosová and V. Davídek, “Design and Implementation of Wave Digital Filters,” *Department of Circuit Theory, Czech Technical University, Prague, Czech Republic September 2001*

[5] Ivan Cohen, Thomas Hélie. Measures and Models of Real Triodes, for the Simulation of Guitar Amplifiers. Société Française d’Acoustique. Acoustics 2012, Apr 2012, Nantes, France. 2012.

[6] Jyri Pakarinen, David T. Yeh, “A Review of Digital Techniques for Modeling Vacuum-Tube Guitar Amplifiers,” Helsinki University of Technology, Helsinki, Finland, Report, August 2009.

[7] James Siegle, “DSP Implementation of a 1961 Fender Champ Amplifier,” B.S. Thesis, Dept. Elect. Eng., Bradley University, Peoria, Illinois, U.S., 2003

[8] Marshall Leach, “SPICE Models for Vacuum-Tube Amplifiers,” Georgia Institute of Technology, Atlanta, Georgia, U.S., Report, March 1995

Appendix A: Matlab Code

```
%% Real-Time Implementation of WDF Triode Stage Amplifier
clc
clear

%% Section A: Initialization
%set gain of audio file
Gain =22;

%obtain info of audio file
fileInfo = audioinfo('Fdeltachord.wav');
Fs = fileInfo.SampleRate; %get sample rate
load('hpf.mat');

%Set frame size
frameSize = 128;

%% Section B: Construction
% set up one port parameters
Rk=Resistor(1e3);
Ck=Capacitor(10e-6,Fs);
Ro=Resistor(1e6);
Co=Capacitor(10e-9,Fs);
ERp=ResistiveVoltageSource(100e3);
Vcc=250;

% port resistances
RS13 = Ro.Rp;
RS12 = Co.Rp;
RS11 = RS13+RS12;
RP13 = ERp.Rp;
RP12 = RS11;
RP11 = 1/((1/RP12)+(1/RP13));

RP23 = Rk.Rp;
RP22 = Ck.Rp;
RP21 = 1/((1/RP22)+(1/RP23));
RS23 = RP21;
```

```
RS22 = RP11;
RS21 = RS23+RS22;

% initialize waves
as1=zeros(1,3);
as2=zeros(1,3);
ap1=zeros(1,3);
ap2=zeros(1,3);

bs1=zeros(1,3);
bs2=zeros(1,3);
bp1=zeros(1,3);
bp2=zeros(1,3);

%set up RT
 fileReader = dsp.AudioFileReader('Filename',fileInfo.Filename,...
    'SamplesPerFrame', frameSize);
 audioReader = audioDeviceReader('SampleRate', fileReader.SampleRate,...
     'Device', 'Scarlett 2i2 USB','SamplesPerFrame', frameSize);
%  audioReader = audioDeviceReader('SampleRate', fileReader.SampleRate,...
%      'Device', 'Pro Tools Aggregate I/O','SamplesPerFrame', frameSize);
%deviceWriter = audioDeviceWriter('SampleRate', fileReader.SampleRate,...
%  'Device', 'Built-in Output');

% deviceWriter = audioDeviceWriter('SampleRate', fileReader.SampleRate,...
%      'Device', 'Pro Tools Aggregate I/O','BufferSize', frameSize);
deviceWriter = audioDeviceWriter('SampleRate', fileReader.SampleRate,...
    'Device', 'Scarlett 2i2 USB','BufferSize', frameSize);

%deviceWriter = audioDeviceWriter('SampleRate', fileReader.SampleRate,...
%     'Device', 'Pro Tools Aggregate I/O');

%% Section C: Audio Stream Loop
ts=1/Fs;        % obtain sampling period
Vg=20;          % initialize Vg
Vk=0;           % initialize Vk
Vpk=0;          % initialize Vpk

%while ~isDone(fileReader)
```

```
while(1)
  % Recieve Data
  %in = fileReader();
  in = audioReader();
  % Process Data
  Vin=in(:,1)/.55; % convert digital back to Voltage
  Vin=Gain*Vin;    % apply gain
  %[out, Vk, Vpk] = TriodeFunc(Vin, Fs, frameSize, Vg, Vk, Vpk);
  for k=1:frameSize
    % 1. Gather inputs
    as1(3)=Ro.get_reflected_wave(bs1(3));
    as1(2)=Co.get_reflected_wave(bs1(2));
    ap1(3)=ERp.get_reflected_wave(bp1(3),Vin(k));
    ap2(3)=Rk.get_reflected_wave(bp2(3));
    ap2(2)=Ck.get_reflected_wave(bp2(2));

    % 2. Wave up
    bs1=SeriesAdaptor(as1,[RS11 RS12 RS13]);
    ap1(2)=bs1(1,1);

    bp1=ParallelAdaptor(ap1,[RP11 RP12 RP13]);
    as2(2)=bp1(1,1);

    bp2=ParallelAdaptor(ap2,[RP11 RP12 RP13]);
    as2(3)=bp2(1,1);

    bs2=SeriesAdaptor(as2,[RS21 RS22 RS23]);
    Ta=bs2(1,1);

    % 3. Root (Triode Valve)
    Vgk=Vg-Vk;
    %    [Vpk, as2(1)] = newton_raphson_solver(Vpk, Vgk, Ta, RS21);
    [Vpk, as2(1)]=Triode(Ta, RS21, Vgk, Vpk);
    %debugVar(k)=as2(1);

    % 4. Wave down
    bs2=SeriesAdaptor(as2,[RS21 RS22 RS23]);
    ap2(1)=bs2(3); %
    ap1(1)=bs2(2); %
```

```
        bp2=ParallelAdaptor(ap2,[RP21 RP22 RP23]);
        %bp2(3)=bp2(3); %
        %bp2(2)=bp2(2); %

        bp1=ParallelAdaptor(ap1,[RP11 RP12 RP13]);
        %bp1(3)=bp1(3); %
        as1(1)=bp1(2); %

        bs1=SeriesAdaptor(as1,[RS21 RS22 RS23]);
        %bs1(2)=bs1(2); %
        %bs1(3)=bs1(3); %

        % 5. Gather outputs
        Vk = Rk.wave_to_voltage();
        output(k) = Ro.wave_to_voltage();

        Ck.set_incident_wave(ap2(2)); % Only works if adaptor incident wave
        Co.set_incident_wave(as1(2)); % Only works if adaptor incident wave

    end
    %out=filter(Hhp,output);
    out=(output')*10000;
    %out = highpass(output,2*pi*10,Fs);
    %plot(out)
    % Write Data
    deviceWriter(out);

end
```