The TubeScreamer TS808: A Digital Model

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Overview

The TS808 was the first of the TubeScreamer guitar pedal series. Through the boosting of mid-frequencies and warm harmonics, the TS808 provides a rich overdrive while preserving the input signal's characteristics. This paper explores how the linear and nonlinear components of the TS808 can be digitally modeled in MATLAB.

Circuit Decomposition: Signal Flow and Subsystems

To analyze the TS808 circuitry and signal flow, I will reference the schematic breakdown as presented on ElectroSmash.com (***Figure 1***).[1] There are six major subsystems in the schematic, but for the purpose of digital modeling I incorporated only the four that affect the digital signal: Input Buffer, Clipping Amp, Tone/Volume Stage, and Output Buffer.

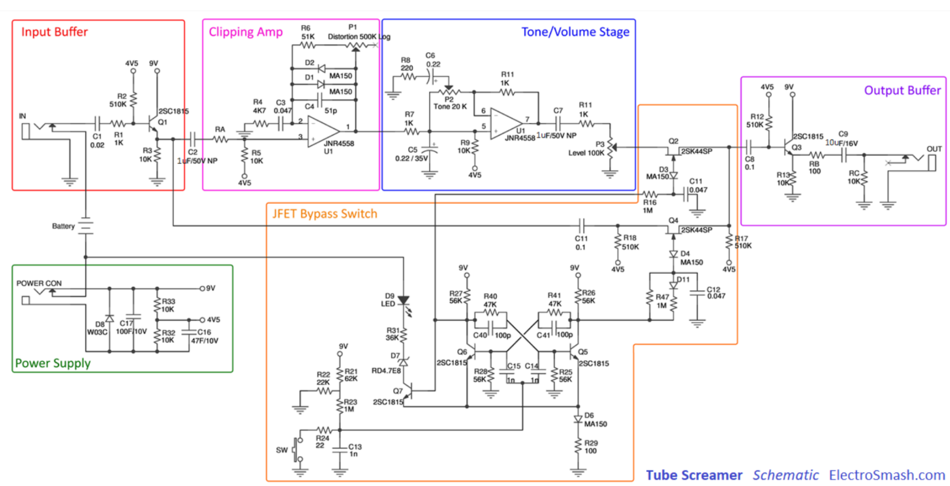


Figure 1

Input Buffer

The first block that the signal encounters is the input buffer, which consists of a first-order RC high-pass filter followed by a unity-gain emitter follower (***Figure 2***). This stage

[1] ElectroSmash - Tube Screamer Circuit Analysis

acts as a DC block filtering any low humming that might be present in the signal and lowers the output impedance for the upcoming blocks in the circuit.

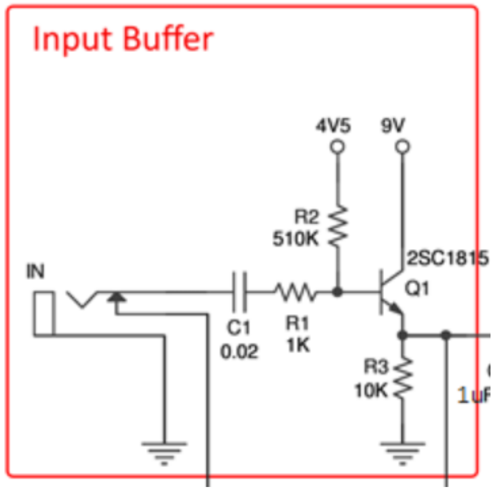


Figure 2

Using an estimated input impedance, I determined the R_{Total} value to be 338 kΩ. This resulted in the low-pass filter holding a cutoff frequency of approximately 23.5 Hz. After filtering, the signal runs through a transistor with a gain scaling value of 0.993 – acting as a unity gain factor. The purpose of this buffer is to lower the output impedance while keeping the high input impedance, ultimately avoiding frequency response alterations when flowing from a high-impedance source to a low-impedance load. The transfer function from the input buffer stage, $H_1(s)$, is as follows:

$$H_1(s) = A_v \frac{RCs}{RCs + 1}$$

$$R = 338k\Omega \qquad C = 20nF \qquad A_v = 0.993$$

Clipping Amp

The second stage of the TS808 circuit is the clipping amp. The combination of diodes, capacitors, and resistors in the feedback loop of the op-amp produces a filtered overdrive that falls dependent on a potentiometer (***Figure 3*****).** The clipping amp block can be broken down further into four subsections: Input Biasing, High-Pass Filter, Clipping Section, and Low-Pass Filter.

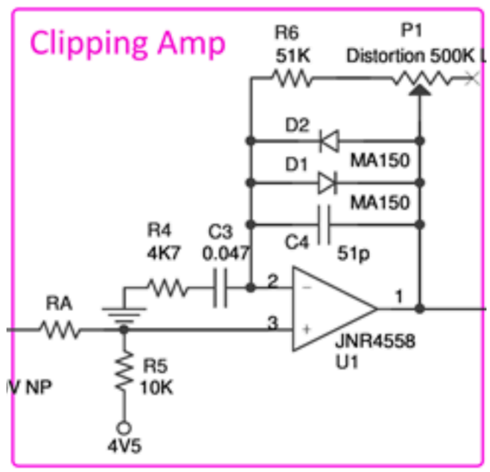


Figure 3

As the circuit runs off of a 9V power supply, the signal must first flow through a biasing section in the clipping amp block resulting in a centered virtual ground around 4.5V rather than 0V. This is the case so that the positive half of the waveform can reach up to 9V and the negative half can reach down to 0V. The next substage is built off of R_4 and C_3 which together form a high-pass filter with a frequency cutoff of approximately 720 Hz. This filter prevents any muddiness that might come from distorting the bass frequencies of the signal. The corresponding transfer function, $H_2(s)$, can be expressed as:

$$H_2(s) = \frac{RCs}{RCs + 1}$$

$$R = 4.7k\Omega \qquad\qquad C = 47nF$$

After the high-pass filter, the signal reaches the nonlinear clipping stage which includes an inverting amplifier followed by parallel diodes with an estimated threshold of 0.5-0.7V. A potentiometer (P_1) controls the amplification amount applied to the signal in series with R_6, with higher values of P_1 equating to more amplification, and vice versa. When the signal from the amplifier exceeds the diode threshold, a soft clipping forms from increased feedback. The last portion of the clipping amp block occurs in the feedback loop with components R_6 and C_4 which create an RC low-pass filter with a cutoff frequency of approximately 60 kHz. Having such a high cutoff frequency due to the 51 pF capacitor maintains the quality of the frequency range while also improving stability in the feedback loop and ensuring control over harmonics. The transfer function describing this final low-pass stage, $H_3(s)$, is:

$$H_3(s) = \frac{1}{RCs + 1}$$

$$R = 51k\Omega \qquad\qquad C = 51pF$$

Tone/Volume Stage

Following the clipping amp block is the tone and volume stage of the TS808. This is the section where low and high frequencies are attenuated or boosted depending on the value of a potentiometer – shaping the frequency content of the signal (***Figure 4***). The volume potentiometer is also located within this stage, just before the signal reaches the final output buffer.

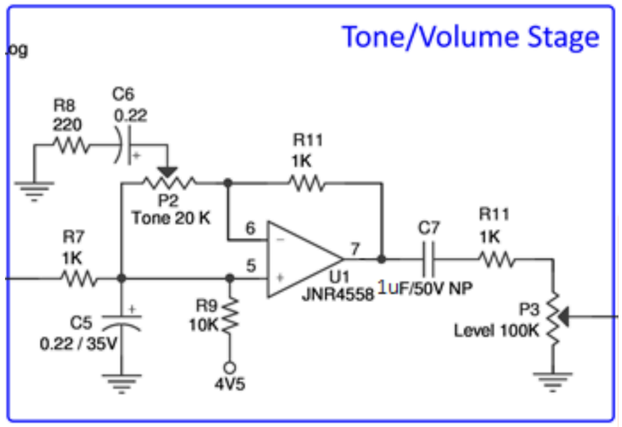


Figure 4

Within the tone and volume network, there are three primary phases in the signal path. The first is a passive low-pass filter using components R_7 and C_5. This filter, with a cutoff frequency of approximately 723 Hz, attenuates the harsh harmonics generated from the previous clipping amp subsystem. The second phase of the tone and volume block is the active mid-frequency filtering circuit which differs depending on the value of the associated potentiometer, P_2. When P_2 holds a value of 0 Ω, R_8 and C_6 hold a stronger connection to the feedback network and act as a second low-pass filter due to increased high-frequency feedback. This results in an added pole to the transfer function meaning a second-order low-pass filter in relation to the low-pass filter from the first phase. Contrastingly, when P_2 = 20 kΩ, the high-frequency feedback is reduced which introduces a zero to the transfer function. This added zero manifests as a band-pass filter in the frequency response with mid-frequency boost. The last phase of the tone and volume network is the volume control, which simply acts as a gain factor after the signal has gone through phases one and two. The tone and volume block can be mathematically represented by this transfer function, $H_4(s)$:

$$H_4(s) = KV\frac{s + w_z}{(s + w_{p1})(s + w_{p2})}$$

$$C_z = 220nF \qquad R_z = 220\Omega$$

$$R_f = 1k\Omega \qquad R_{load} = 10k\Omega \qquad R_{pot} = 20k\Omega$$

$$R_l = \begin{cases} TR_{pot}, & TR_{pot} \geq 1 \\ 1, & TR_{pot} < 1 \end{cases} \qquad R_r = \begin{cases} (1-T)R_{pot}, & TR_{pot} \geq 1 \\ 1, & TR_{pot} < 1 \end{cases}$$

$$w_{p1} = \frac{1}{C_z(R_z + R_f)} \qquad w_{p2} = \frac{1}{C_z R_{load}} \qquad w_z = \frac{1}{C_z(R_z + R_l)}$$

$$K = \frac{R_{load} + R_f}{R_{load}} \qquad V = Volume/100 \qquad T = Tone/100$$

$$Tone \in [0, 100] \qquad Volume \in [0, 100]$$

Output Buffer

The final block of the TS808 circuit breakdown is the output buffer (***Figure 5***). This stage is almost identical to the input buffer with a DC bias at 4.5V, an emitter follower to lower output impedance, and a high-pass filter to rid any low-frequency humming noise.

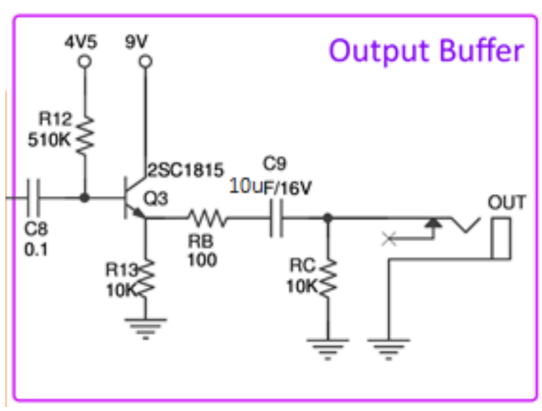


Figure 5

Similar to the input buffer, the output buffer works at a unity-gain factor of approximately 1. As the signal flows through the transistor, the output impedance is lowered allowing for drive capabilities and preparing for the next stages of an amplifier. The component R_B provides stability in the circuit by slightly reducing the gain. Once the signal passes through C_9, DC bias is effectively removed and the AC signal is centered around 0V. C_9 also works in series with R_C to form a high-pass filter with a cutoff frequency of around 16 Hz. The transfer function for the final output buffer stage, $H_5(s)$, can be represented by:

$$H_5(s) = \frac{RCs}{RCs+1}$$

$$R = 10k\Omega \qquad C = 10uF$$

Analog-to-Digital Conversion

After acquiring the five transfer functions, I then continued my modelling process by converting the subsystem representations to digital models using the bilinear transform, or bilinear() command in MATLAB. From there, I extracted the difference equation for each digital model.

H_1(s)

$$H_1(s) = A_v \frac{RCs}{RCs+1}$$

$$H_1(z) = \frac{0.9915 - 0.9915z^{-1}}{1 - 0.9969z^{-1}}$$

$$y_1[n] = 0.9915x[n]-0.9915x[n-1]+0.9969y[n-1]$$

H_2(s)

$$H_2(s) = \frac{RCs}{RCs+1}$$

$$H_2(z) = \frac{0.9550 - 0.9550z^{-1}}{1 - 0.9099z^{-1}}$$

$$y_2[n] = 0.9550x[n]-0.9550x[n-1]+0.9099y[n-1]$$

H_3(s)

$$H_3(s) = \frac{1}{RCs+1}$$

$$H_3(z) = \frac{0.8002 + 0.8002z^{-1}}{1 + 0.6004z^{-1}}$$

$$y_3[n] = 0.8002x[n]+0.8002x[n-1]-0.6004y[n-1]$$

H_4(s)

$$H_4(s) = KV\frac{s+w_z}{(s+w_{p1})(s+w_{p2})}$$

$$H_4(z) = \frac{1.0e^{-4}(0.1054 + 0.0018z^{-1} - 0.1036z^{-2})}{1 - 1.8220z^{-1} + 0.8253z^{-2}}$$

$$y_4[n] = 1.054e^{-5}x[n]+1.8e^{-7}x[n-1]-1.036e^{-5}x[n-2]+1.8220y[n-1]-0.8253y[n-2]$$

H_5(s)

$$H_5(s) = \frac{RCs}{RCs+1}$$

$$H_5(z) = \frac{0.9999 - 0.9999z^{-1}}{1 - 0.9998z^{-1}}$$

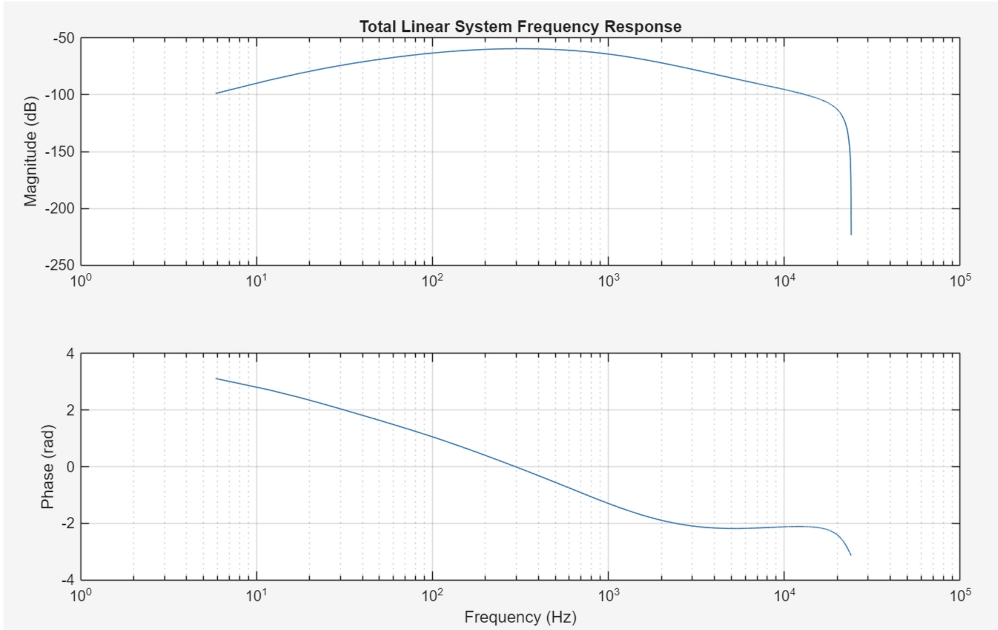
$$y_5[n] = 0.9999x[n]-0.9999x[n-1]+0.9998y[n-1]$$

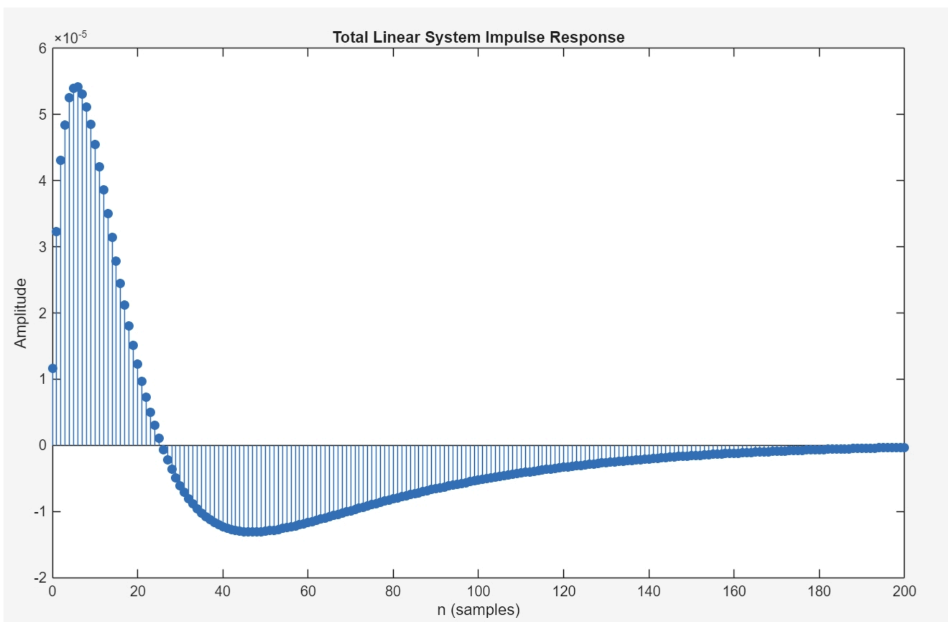
MATLAB Implementation

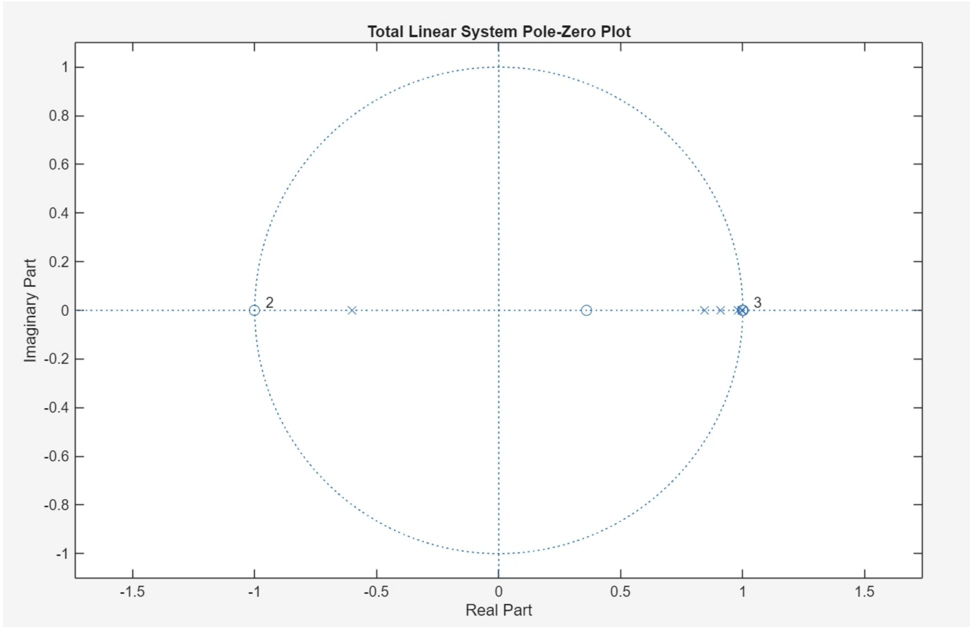
By implementing my calculations into MATLAB, I was able to demonstrate the frequency response, impulse response, and pole-zero plot at each linear block including the input buffer, the tone/volume stage, and the output buffer.

Plots with Parameters 1

Drive = 0, Tone = 0, Volume = 100

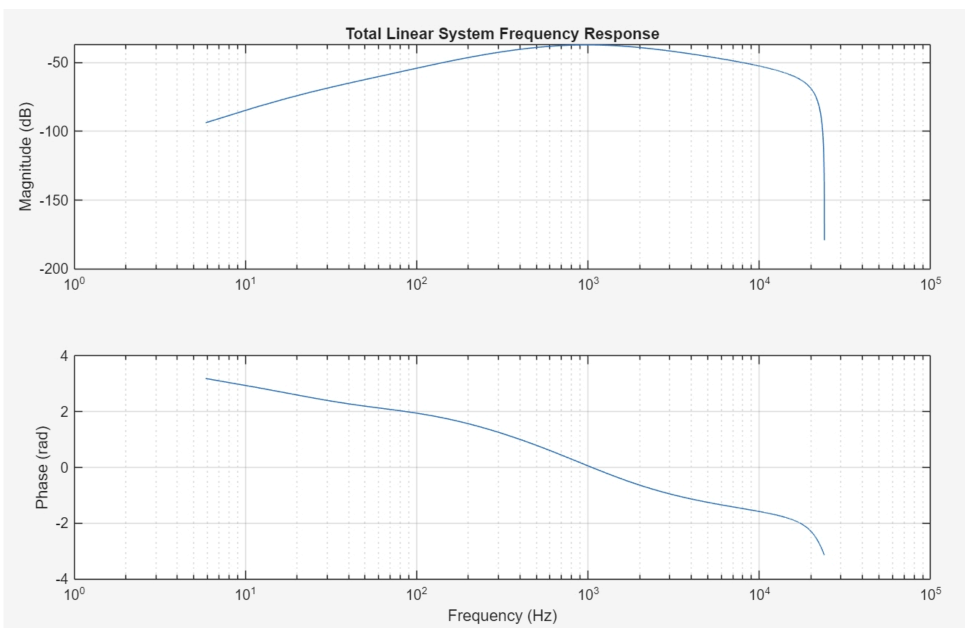

Total Linear System Frequency Response
Magnitude (dB)
-50
-100
-150
-200
-250
Phase (rad)
4
2
0
-2
-4
10^0
10^1
10^2
10^3
10^4
10^5
Frequency (Hz)

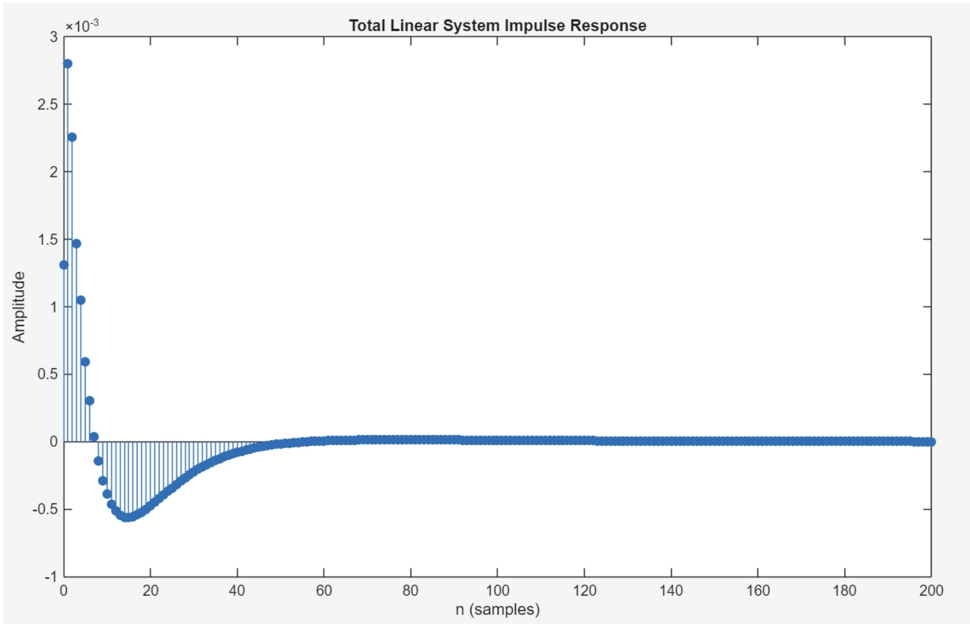

Total Linear System Impulse Response
$\times 10^{-5}$
Amplitude
6
5
4
3
2
1
0
-1
-2
0
20
40
60
80
100
120
140
160
180
200
n (samples)

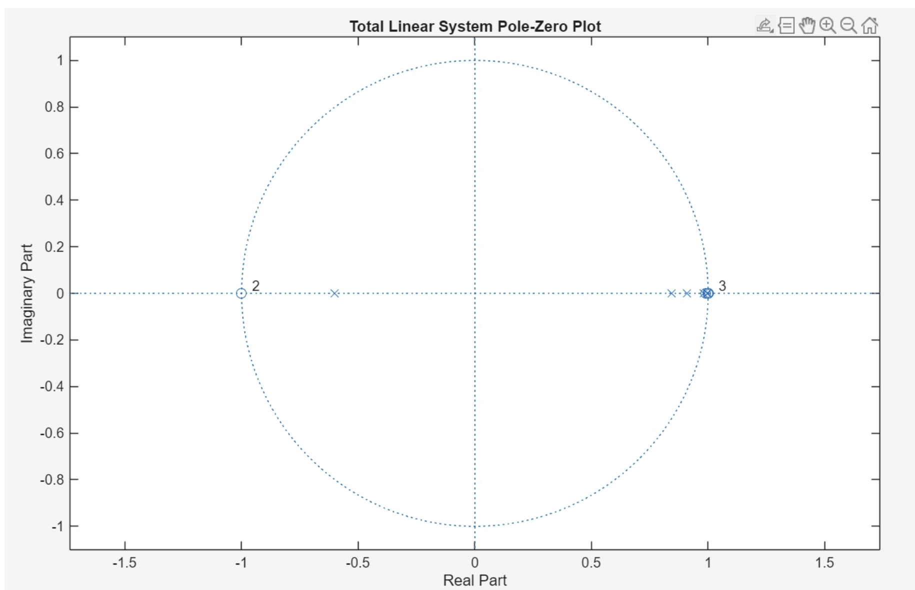


Plots with Parameters 2

Drive = 0, Tone = 100, Volume = 100

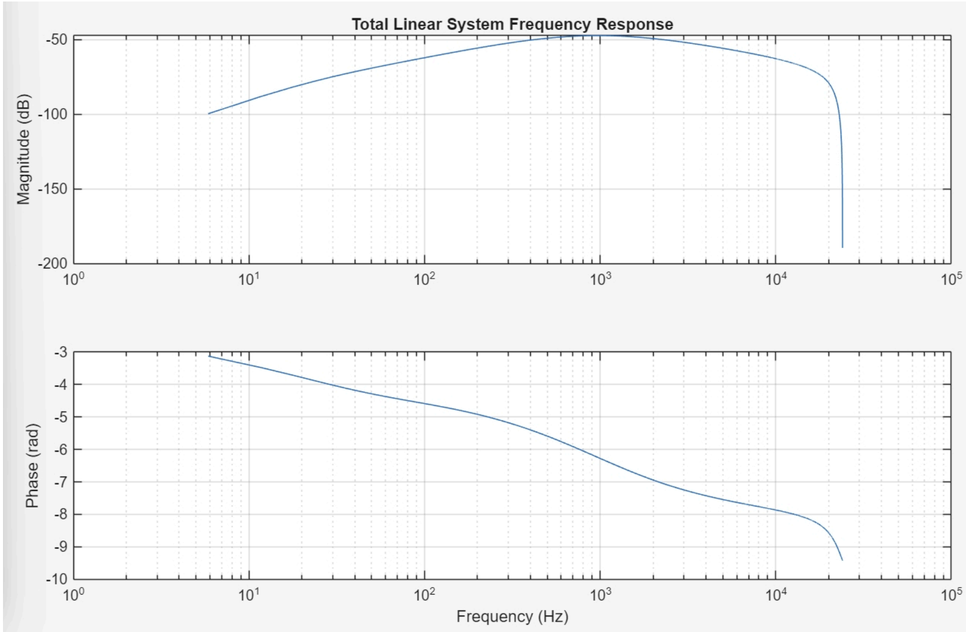


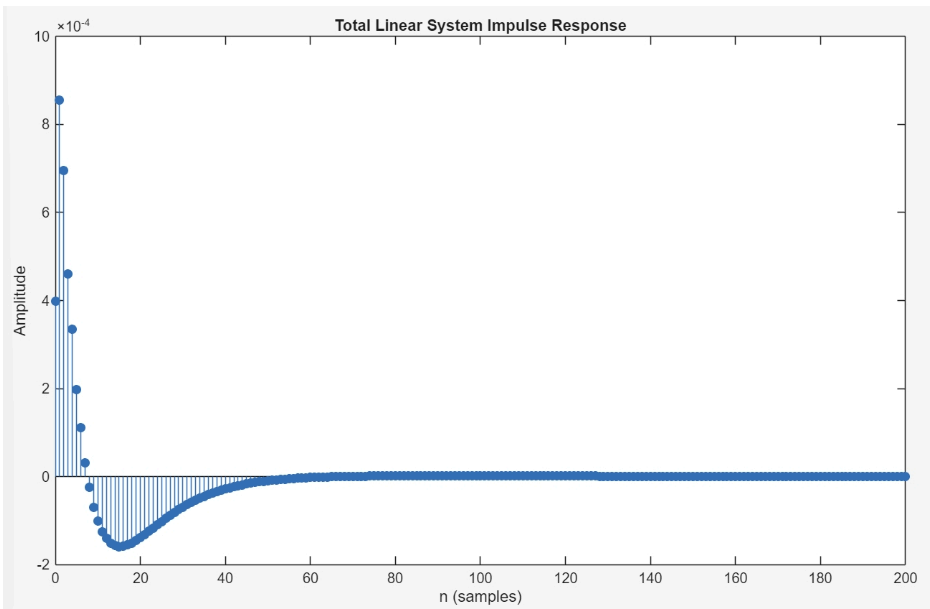


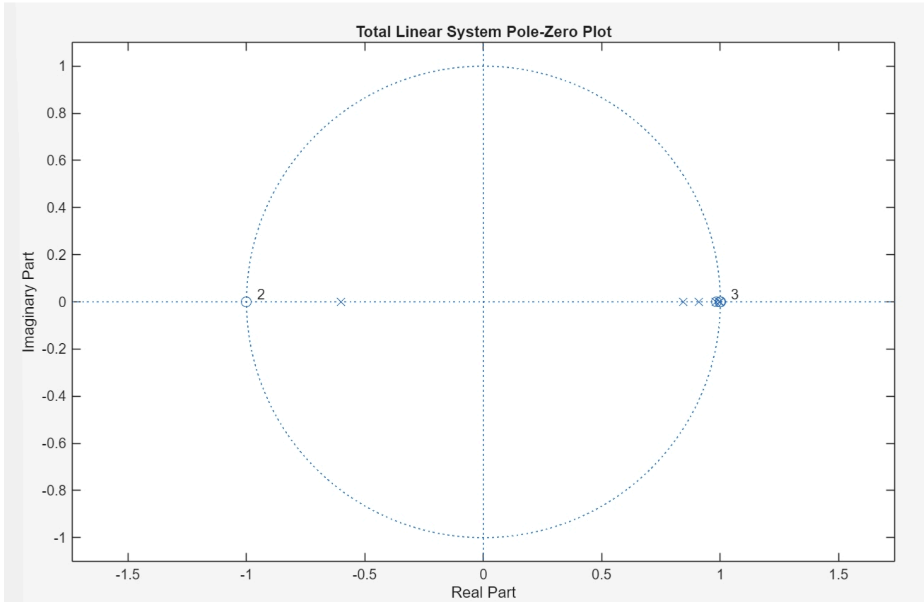


Plots with Parameters 3

Drive = 80, Tone = 60, Volume = 70


Total Linear System Frequency Response
Magnitude (dB)
-50
-100
-150
-200
10^0 10^1 10^2 10^3 10^4 10^5
Phase (rad)
-3
-4
-5
-6
-7
-8
-9
-10
Frequency (Hz)


Total Linear System Impulse Response
$\times 10^{-4}$
Amplitude
10
8
6
4
2
0
-2
0 20 40 60 80 100 120 140 160 180 200
n (samples)



Clipping Stage Code

To represent the nonlinear portion, the clipping amp stage, I utilized the tanh() function in MATLAB. This allowed for soft clipping and flexibility with the drive parameter variable simultaneously:

```
drive_alpha = Drive / 100;
gain = 1 + 9*(drive_alpha^2);
x_gain = gain .* x_drivehp; Vt = 0.3;
x_nl = Vt * tanh(x_gain / Vt);
x_clip = (1 - drive_alpha) * x_hp + drive_alpha * x_nl;
```

Sources

[**ElectroSmash - Tube Screamer Circuit Analysis**]