Design Problem 1

Design an amplitude modulator using anyone of the methods covered in the class. The baseband signal is an audio signal m(t), which can be generated from an actual recorded clip of your favorite music.

The carrier frequency should be in X-band. The modulated signal should have an average power of 30 dBm with a modulation depth of +/-5 dB. Download the data sheets of the components from the manufacturer or distributor web sites to design your modulator circuit to meet the specifications.

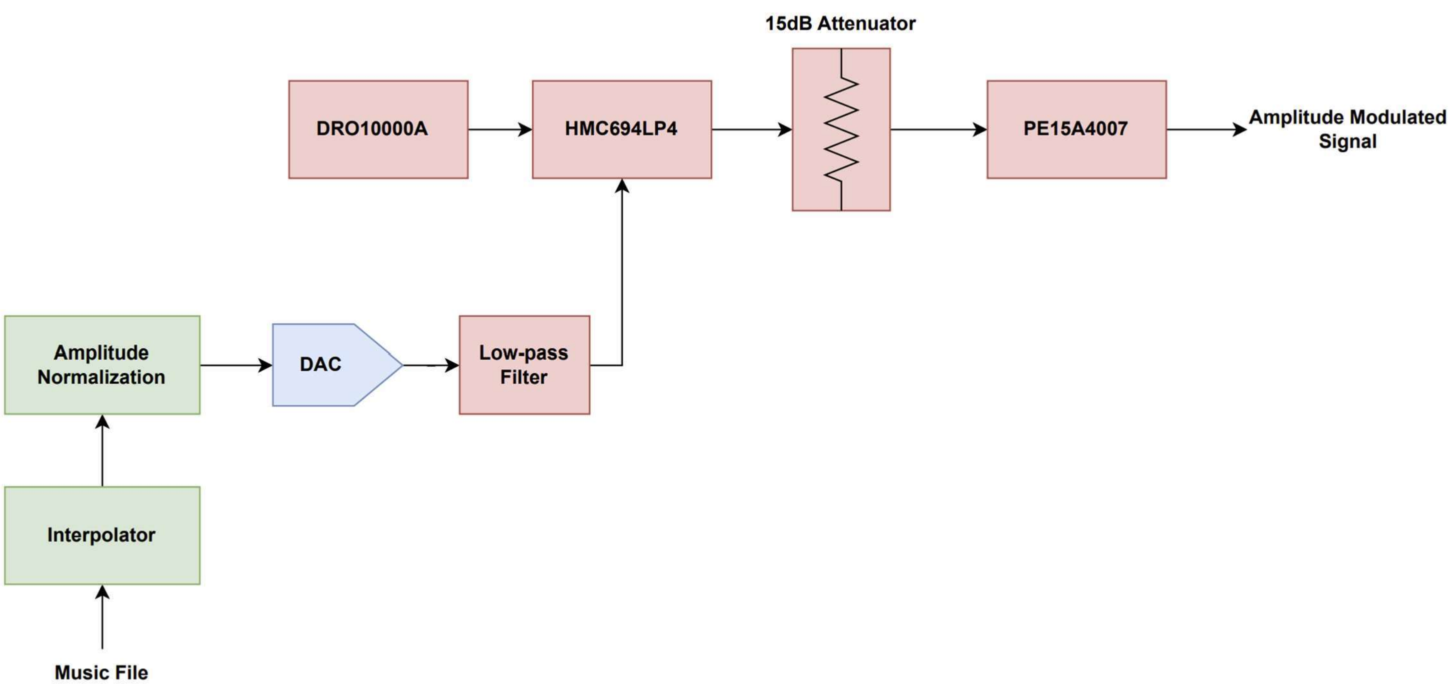
The deliverable is a written report, which should contain a description of the design, complete circuit schematic, analytic description, circuit simulation results that confirm the required objectives. Circuit simulation can be done using CAD tools or MATLAB

Introduction

Our approach to the design problem involves using a digital-to-analog converter (DAC) to synthesize an analog waveform from the digital samples contained in our music file. This analog signal can then be used to modulate the gain of a variable gain amplifier, which amplifies a carrier signal in the X band. In order to meet the modulation depth requirement, the peak-to-peak amplitude of the modulating signal must remain within the range where the amplifier provides a total gain swing of 10dB to the carrier. To satisfy the average power requirement, a combination of attenuators and fixed-gain amplifiers can be cascaded to adjust the average power to 30dBm. The following components will be used in this design:

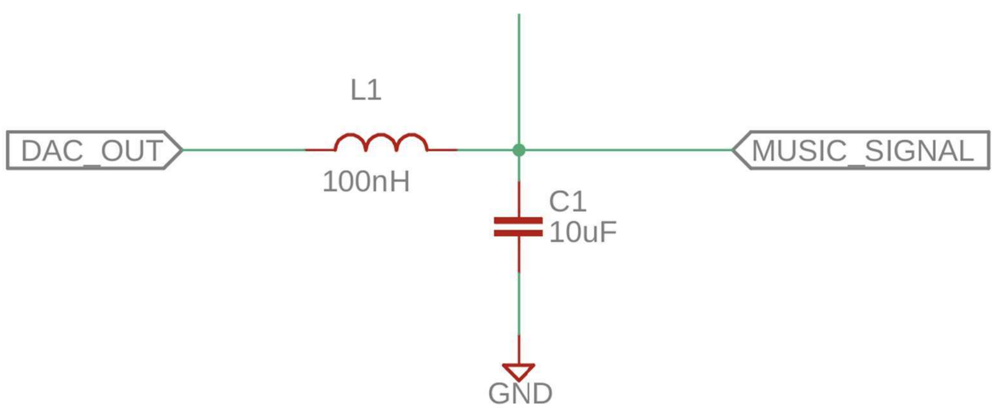
- **DRO10000A** – Voltage-Controlled Oscillator: Output power of 0dBm and output frequency of 10GHz
- **HMC694LP4** – GaAs MMIC Analog Variable Gain Amplifier: 6 – 17 GHz and 23dB gain control with a maximum output power of 22dBm
- **PE15A4007** – X-band Coaxial Power Amplifier: 8.5 – 11GHz and 30dB gain with a maximum output power of 36dBm

A block diagram of our design is show below:



Digital Processing of the music signal

Since our music signal was sampled at a rate of 44.1KHz, which is standard for audio signals, we know that the original signal is band-limited to the Nyquist-frequency of 22.05KHz. Therefore, we can increase our sample rate by inserting zeroes in between samples and convolving the sequence with a Sinc function. Increasing the sample rate digitally allows more samples of the music signal to be sent to the digital-to-analog converter (DAC), giving us a more accurate reconstruction. Having more samples also allows us to use a low-pass filter with a higher cut-off frequency to interpolate between the samples at the output of the DAC. Before the samples are sent to the DAC, we must limit the amplitude of the modulating signal to ensure that it does not cause the gain of the variable gain amplifier (VGA) to exceed the 10dB modulation depth requirement. Additionally, we must add a dc bias to the music signal so that the control voltage is biased within the linear region of the variable gain amplifier. Since most DACs can output both positive and negative voltages, we can bias the signal digitally before sending it to the DAC, with no need for an external clamping circuit.



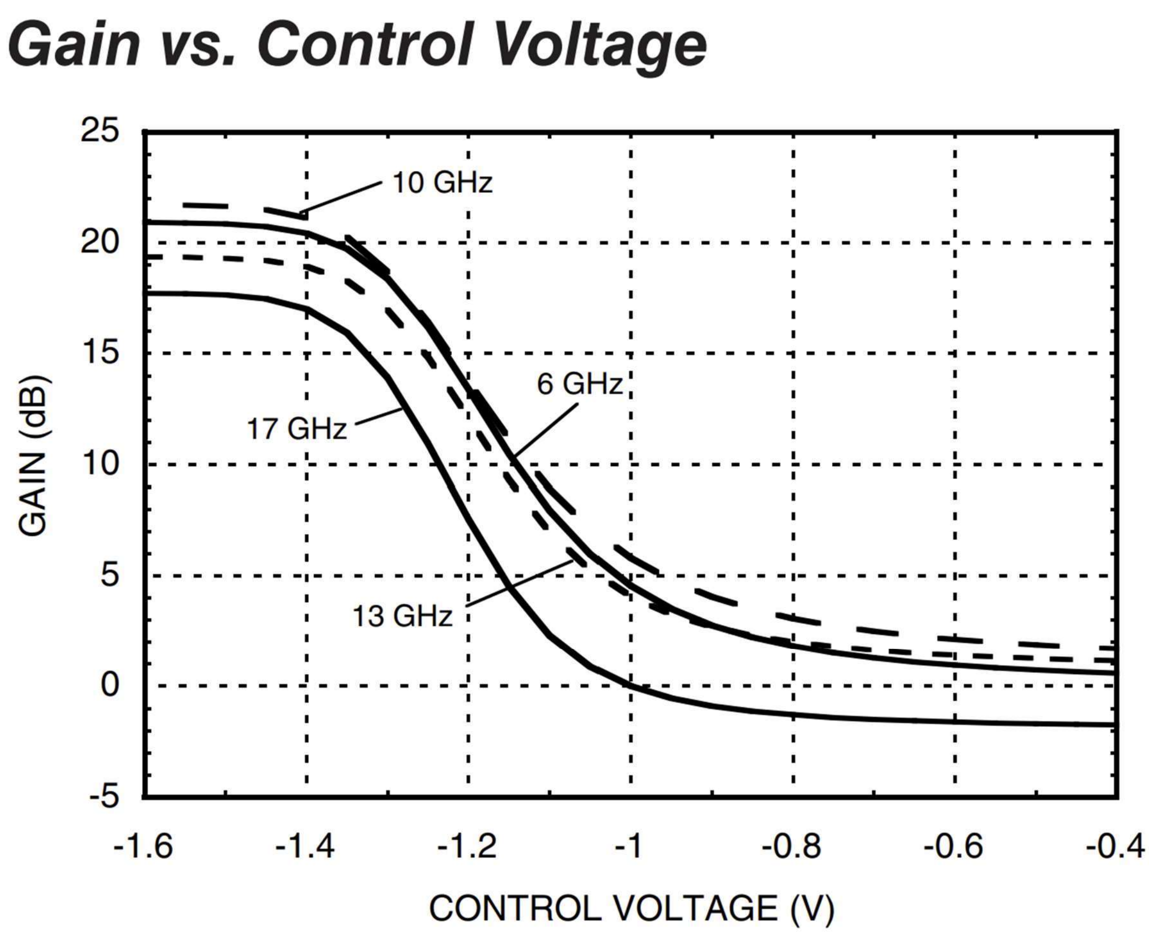
An LC low-pass filter is used to filter out the harmonics of the DAC output. An LC filter is chosen because it is built with lossless reactive components. Additionally, the presence of 2 reactive components gives the filter a -40dB/decade slope above the cut-off frequency. The cut-off frequency of this filter is:

$$f_c = \frac{1}{2\pi\sqrt{LC}} \cong 160kHz$$

By increasing the sampling rate digitally, before sending the samples to the DAC, we can use a low-pass filter with smaller inductors and capacitors and a higher cut-off frequency.

HMC694LP4 Variable Gain Amplifier

The datasheet of the HMC694LP4 includes a graph of the gain in dB versus the control voltage at different frequencies. In order to meet the design requirements, the gain of the VGA must be a linear function of the control voltage which is the modulating signal. The X-band is the range of frequencies between 8-12 GHz, therefore, we will be using a carrier frequency of 10GHz.



Since we require a modulation depth of $\pm$5dB, we would like to find a region on the graph where gain is a linear function of the control voltage over a range which covers at least 10dB of gain. We can see graphically that the gain of the HMC694LP4 at 10GHz has a slope of $\alpha = 50\text{dB/V}$ and is centered at the point $V_0 = -1.225V$ and $G_0 = 15dB$ in the linear region. If the modulating signal m(t) is biased at V_0 and its amplitude is bounded between -1.125V and -1.325V, then the gain of the amplifier is:

$$G[dB] = G_0 + \alpha[m(t) - V_0] \rightarrow \boldsymbol{G[dB] = 15 + 50[m(t) + 1.225]}$$

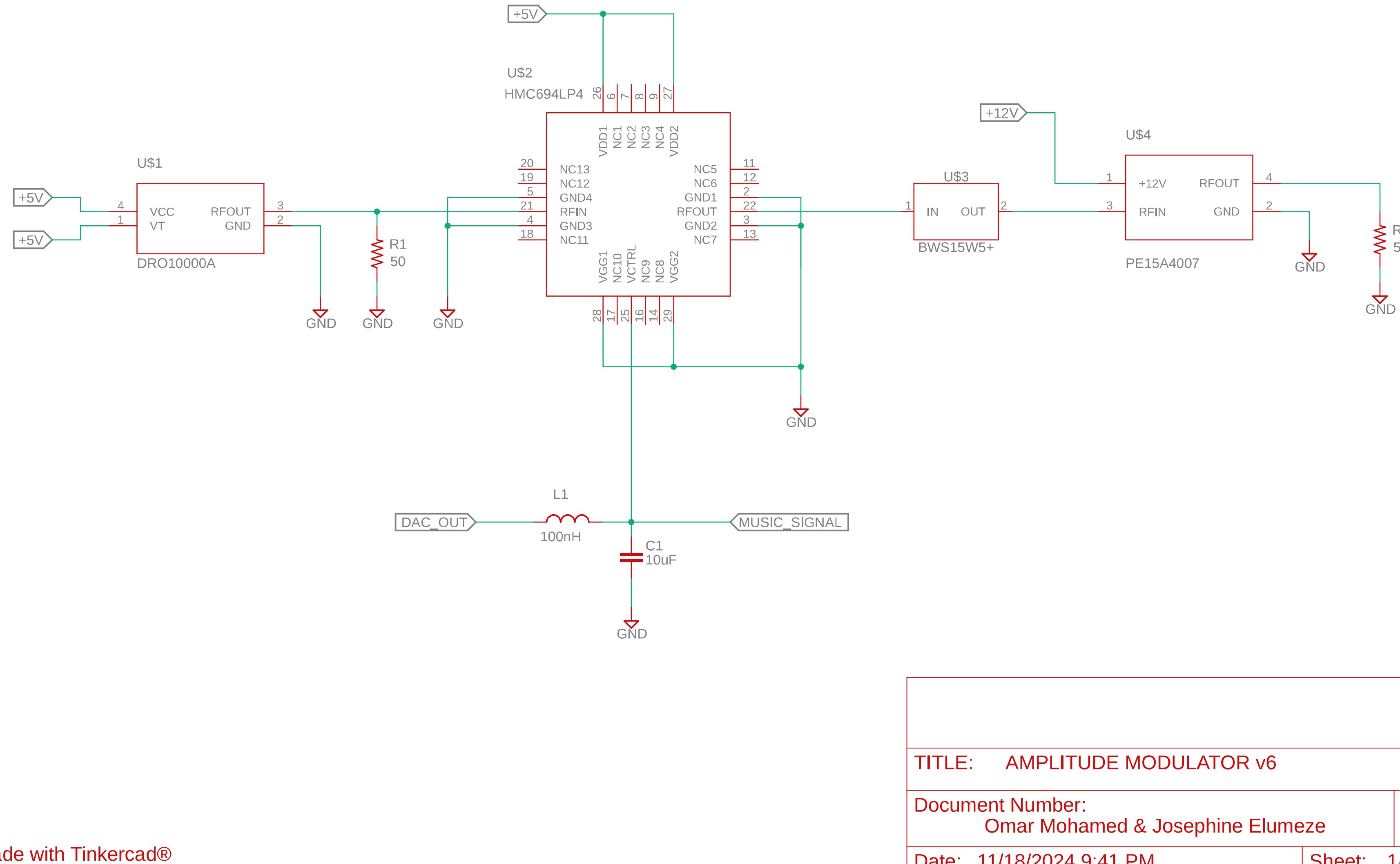
RF Signal Chain

The X-band is the range of frequencies between 8-12 GHz and the source of the carrier frequency is the DRO10000A voltage-controlled oscillator which outputs 0dBm of power with a frequency of 10 GHz. If this carrier signal is sent to the HMC694LP4 and amplified with the gain G from the previous section, then the power will vary between 10dBm and 20dBm with an average power of 15dBm. The carrier signal from the oscillator has successfully been modulated with the required modulation depth of ±5dBm, the average power of the modulated signal must now be increased to 30dBm to meet the design requirements. This is achieved by cascading a 15dB attenuator with a 30dB power amplifier resulting in a net amplification of 15dB. The attenuator scales the signal down to between -5dBm and +5dBm and then the power amplifier scales it up to between +25dBm and +35dBm with an average value of 30dBm. The PE15A4007 power amplifier can output up to 4W which is equivalent to 36dB, enough to meet our maximum power requirement of 35dBm. This circuit topology will meet the design requirements of ±5dB modulation depth with an average signal power of 30dBm.

Bill of Materials

Manufacturer	Manufacturer Part Number	Description
Z Communications	DRO1000A	10GHz Oscillator
Analog Devices	HMC694LP4	Variable Gain Amplifier
Minicircuits	BWS15W5+	15dB Attenuator
Pasternack	PE15A4007	30dB RF Power Amplifier
InspiredLED	12VDC Class 2 Plug-In	+12V AC/DC Wall Mount Adapter
Texas Instruments	TPS54335A	+12V to +5V DC/DC Converter

The DRO1000A and HMC694LP4 require a +5VDC power supply and the PE15A4007 requires a +12VDC power supply. We will be using a +12V AC adapter as our DC power supply for the PE15A4007 and the +12V from this power supply will be stepped down to +5VDC with a DC-DC converter



TITLE: AMPLITUDE MODULATOR v6

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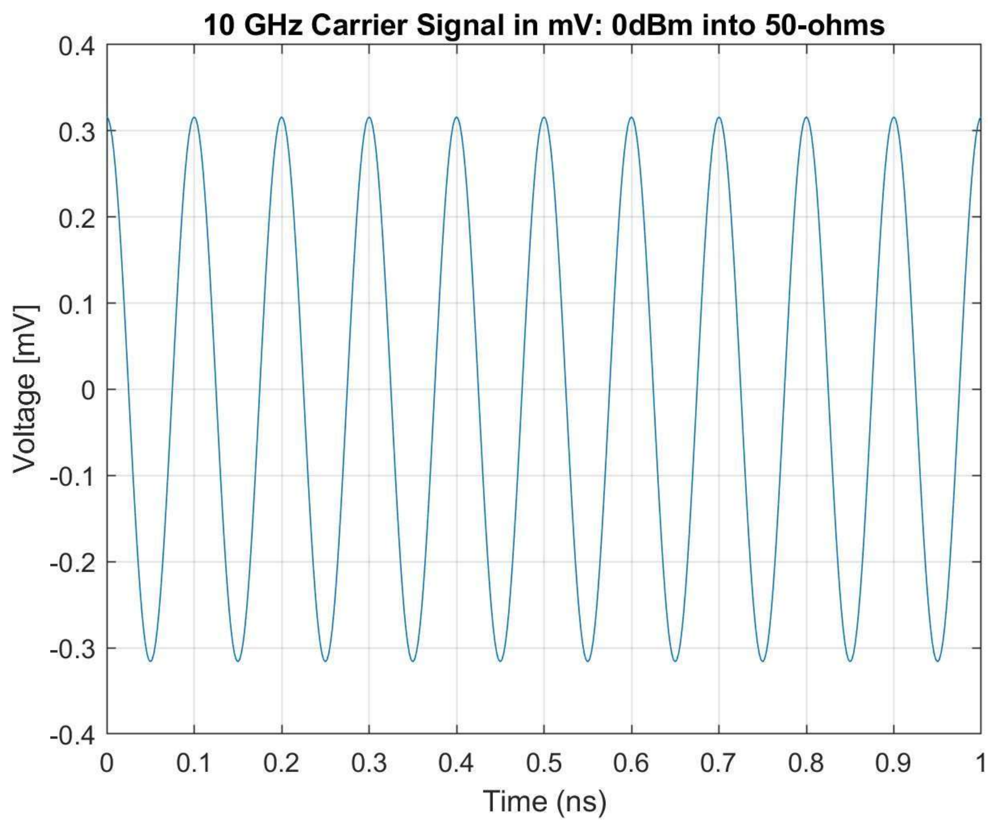
Simulation Waveforms – Oscillator Output

The voltage-controlled oscillator used to generate the carrier signal has 0dBm of power and a frequency of 10GHz. This power is dissipated in a 50Ω resistor, therefore, its amplitude in millivolts is:

$$0dBm = 10\log\left(\frac{A_{osc}^2}{2R}\right) + 30 \rightarrow A_{osc} = \sqrt{0.1}\,mV$$

Shown below is a MATLAB plot of the 0dBm carrier signal in millivolts:

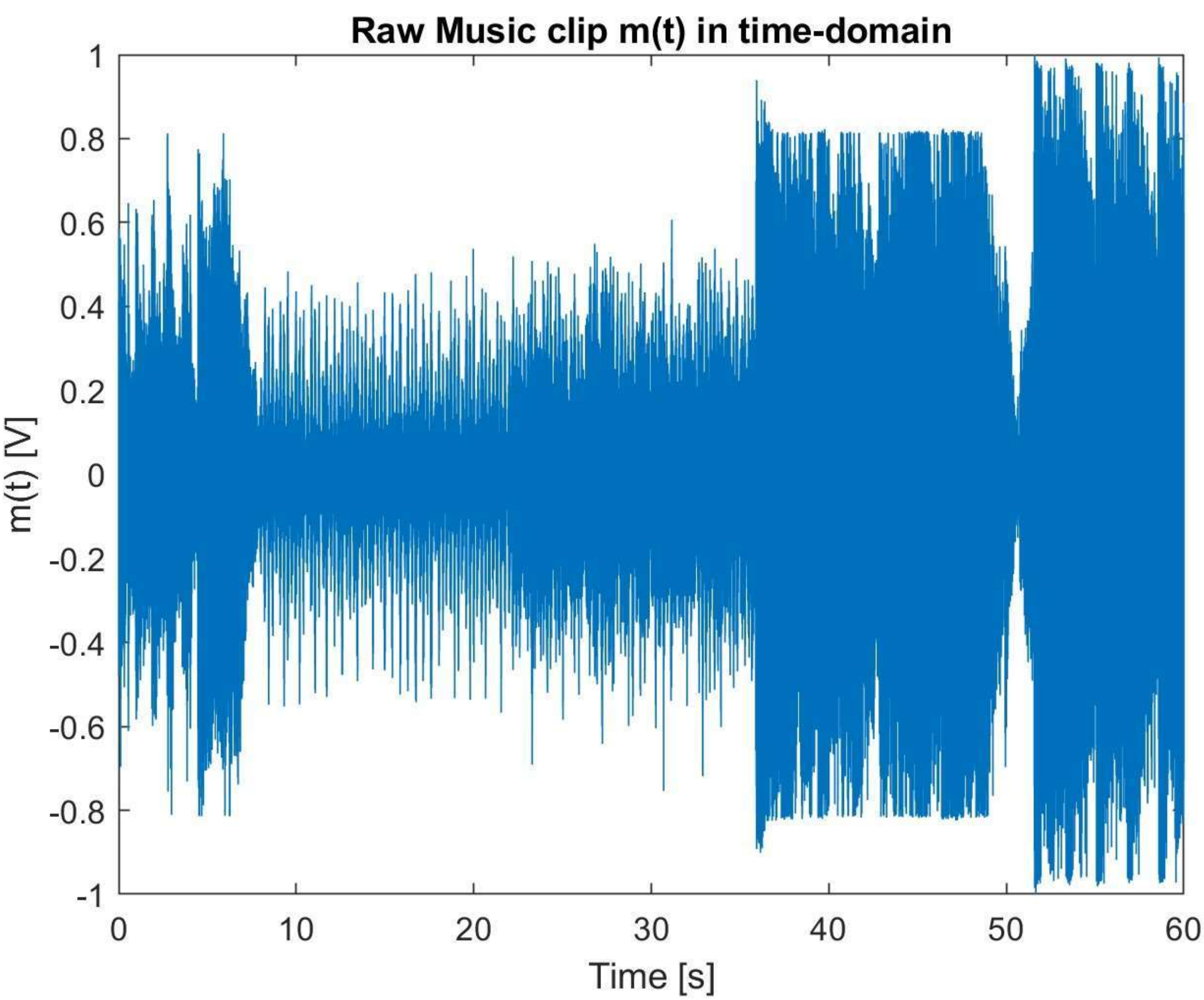


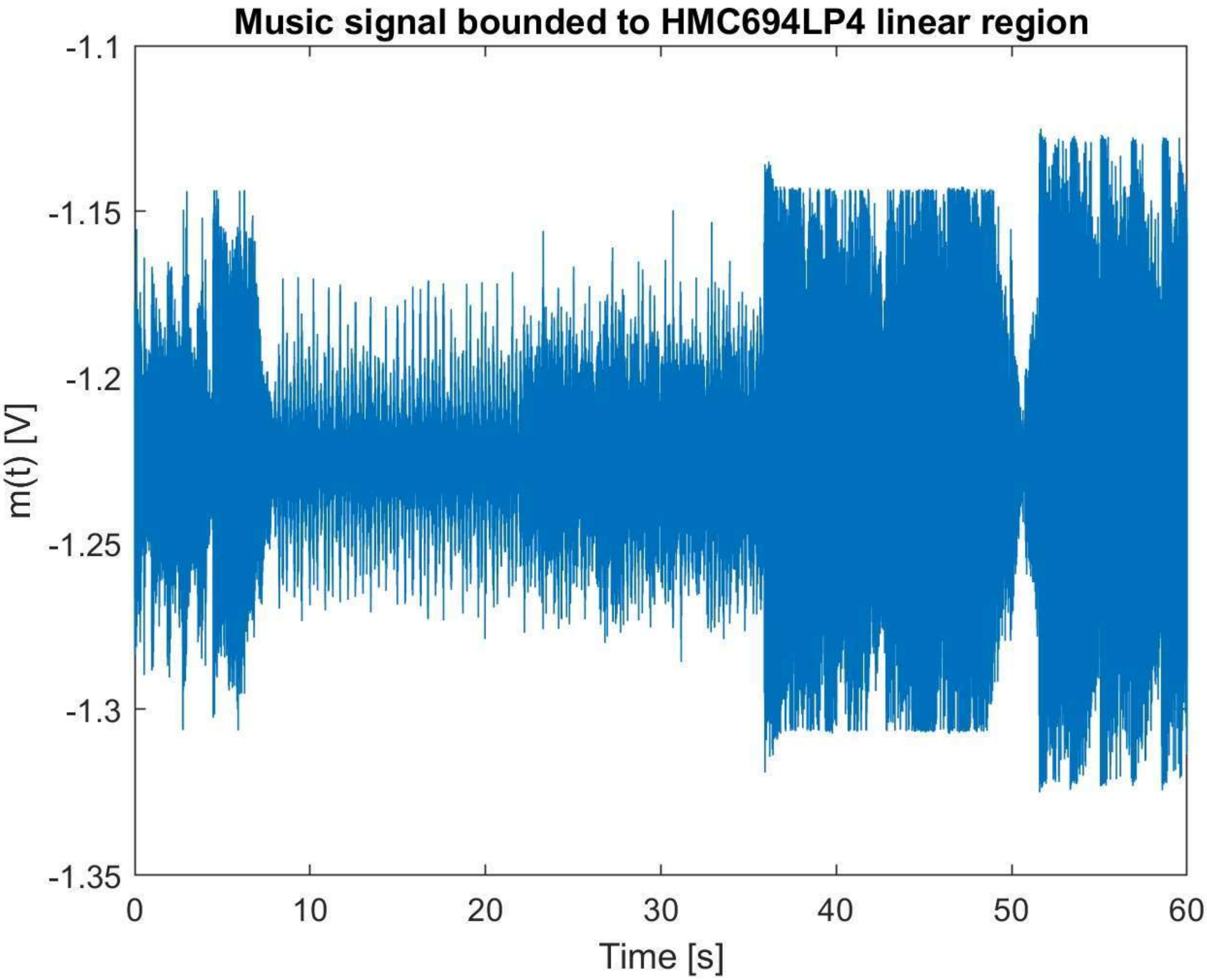
Simulation Waveforms – Music Clip

Interpolation is used to increase the sample rate of the modulating signal read from the music file. The music signal is an ac-signal, therefore, it is centered a 0V. The maximum value of the signal is calculated in MATLAB and then the amplitude is normalized to fit within the linear region of the amplifier:

$$A_{normalized} = A_{max}\left(\frac{V_{max} - V_{min}}{2A_{max}}\right)$$

The new signal is then biased at $V_0 = -1.225V$ so that the control voltage can now vary within the amplifier's linear region. Shown below is a MATLAB plot of the raw music signal read from the file and the music signal when bounded to within the amplifier's linear region:





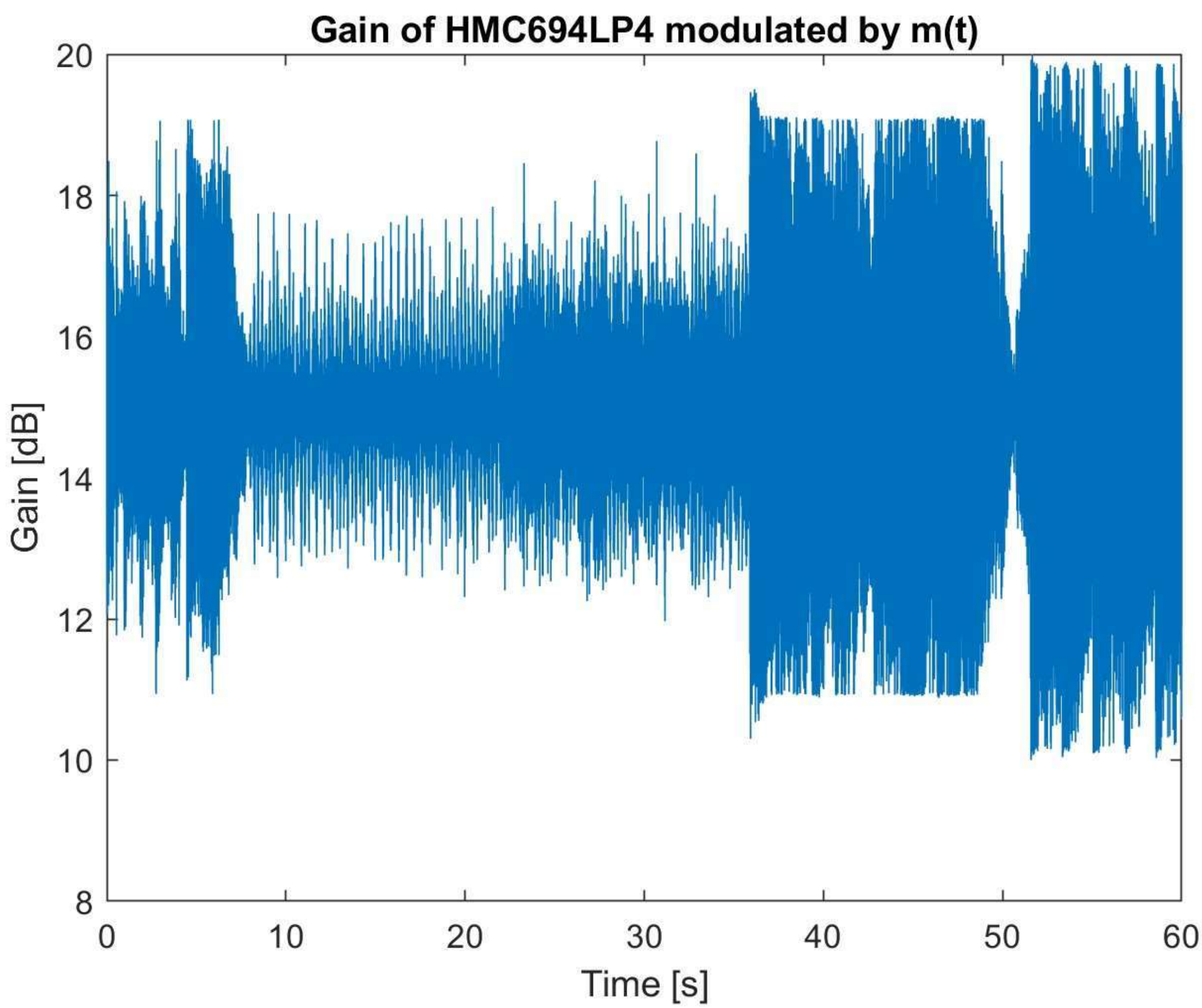
Music signal bounded to HMC694LP4 linear region
-1.1
-1.15
-1.2
-1.25
-1.3
-1.35
m(t) [V]
0
10
20
30
40
50
60
Time [s]

Simulation Waveforms – Gain of the HMC694LP4

Now that the control voltage of the amplifier is bounded to its linear region, the gain will vary between a maximum value of +20dB, a minimum value of +10dB, and an average value of +15dB. The gain in the linear region is defined by the following equation:

$$G[dB] = 15 + 50[m(t) + 1.225]$$

Shown below is a MATLAB plot of the gain of the HMC694LP4 modulated by the music signal that is bounded to its linear region:

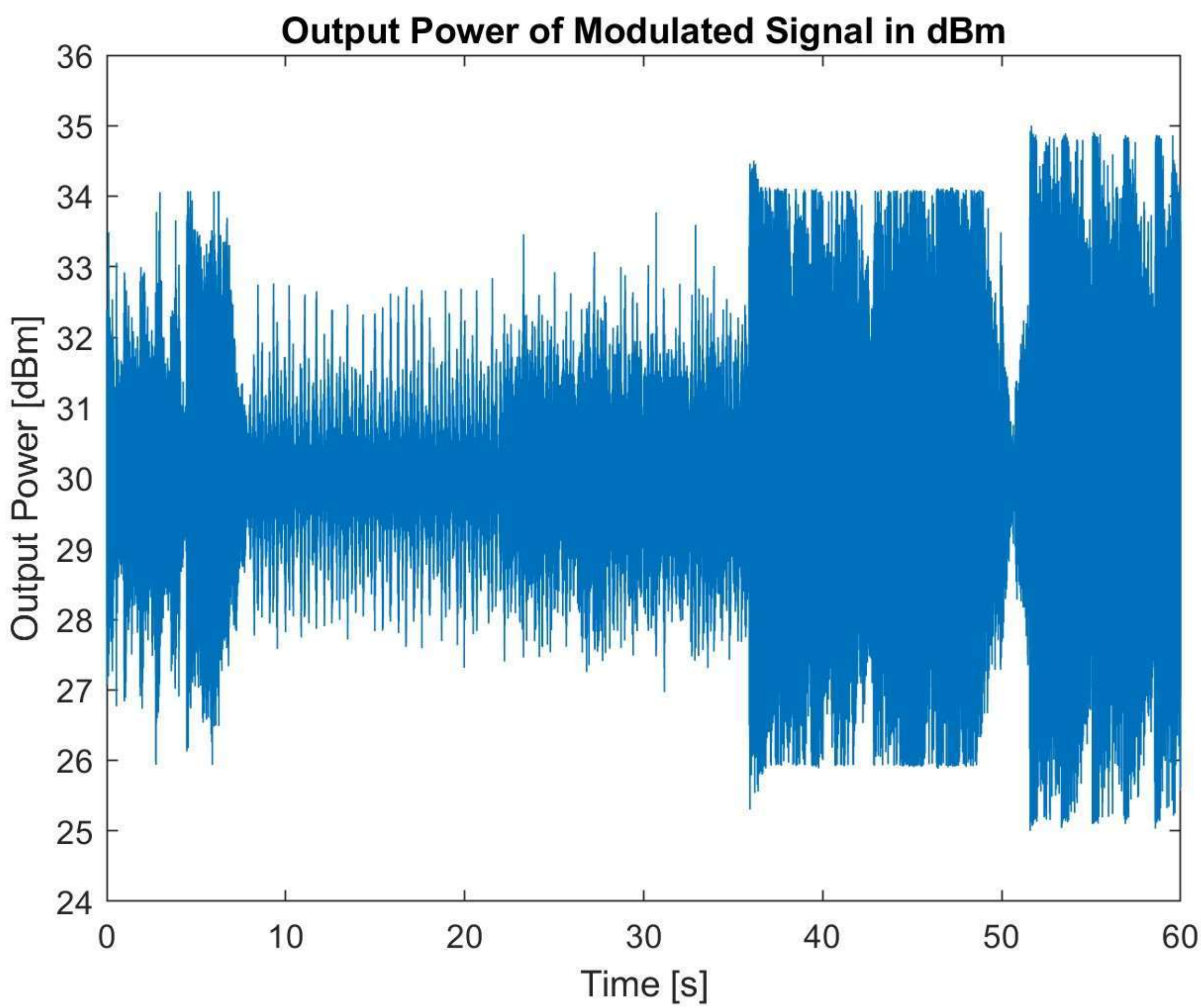


Simulation Waveforms – Attenuator and Power Amplifier

The carrier signal coming out of the HMC694LP4 has an average power of 15dBm with a maximum power of 20dBm and a minimum power of 10dBm. The attenuators reduces the carrier power by 15dBm, and the power amplifier increases the power by 30dBm for a net power increase of 15dBm. The output power is now:

$$P_{out}[dBm] = 30 + 5m_{raw}(t)$$

Where m(t) has a maximum value of +5dBm and a minimum value of -5dBm, resulting in an average power of 30dBm, a minimum power of 25dBm, and a maximum power of 35dBm. Shown below is a MATLAB plot of the output power of the amplitude modulator in dBm.



Simulation Waveforms – Output Voltage

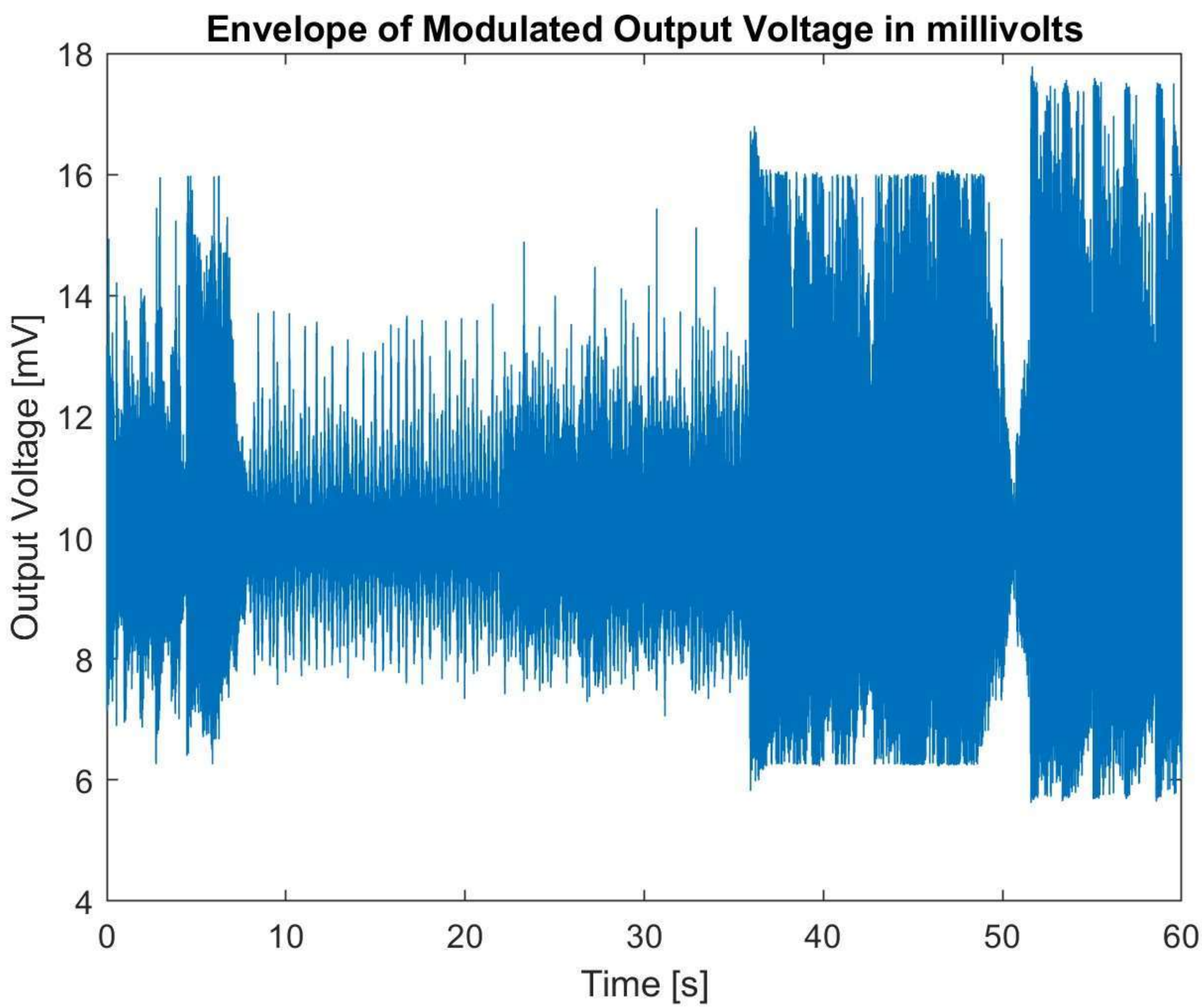
The modulated signal has an average power of 30dBm, a minimum power of 25dBm, and a maximum power of 35dBm. If this power is dissipated in a 50Ω resistor, the envelope function of the output voltage in millivolts will be:

$$V_{out\ envelope} = \sqrt{2R * 10^{\left(\frac{1}{2}\right)\left(\frac{(P_{out}[dBm]-30)}{5}\right)}}\ mV$$

The output voltage of the amplitude modulator will be:

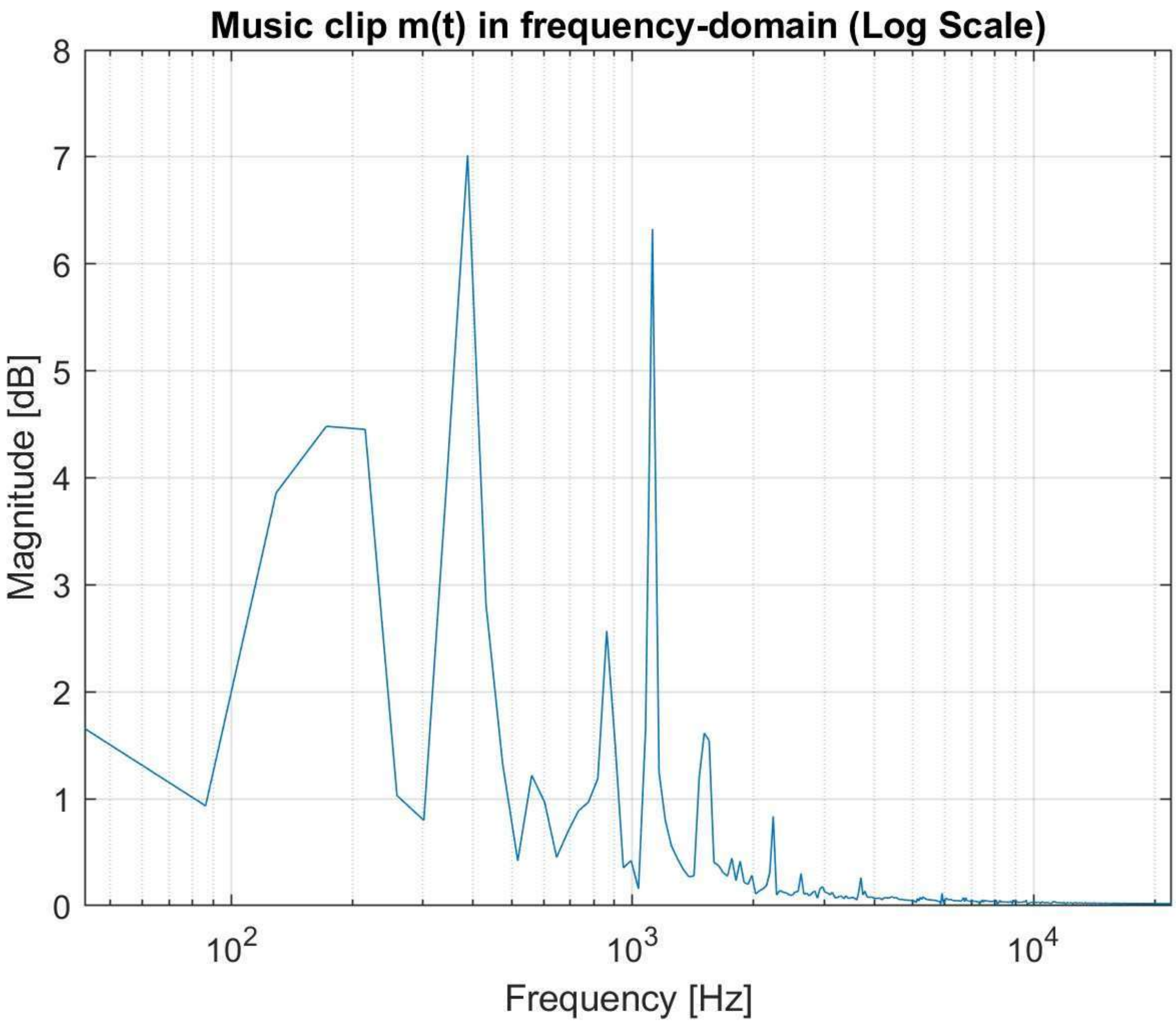
$$V_{out} = \mathrm{A}\cos(\omega_c t)\, V_{out\ envelope}$$

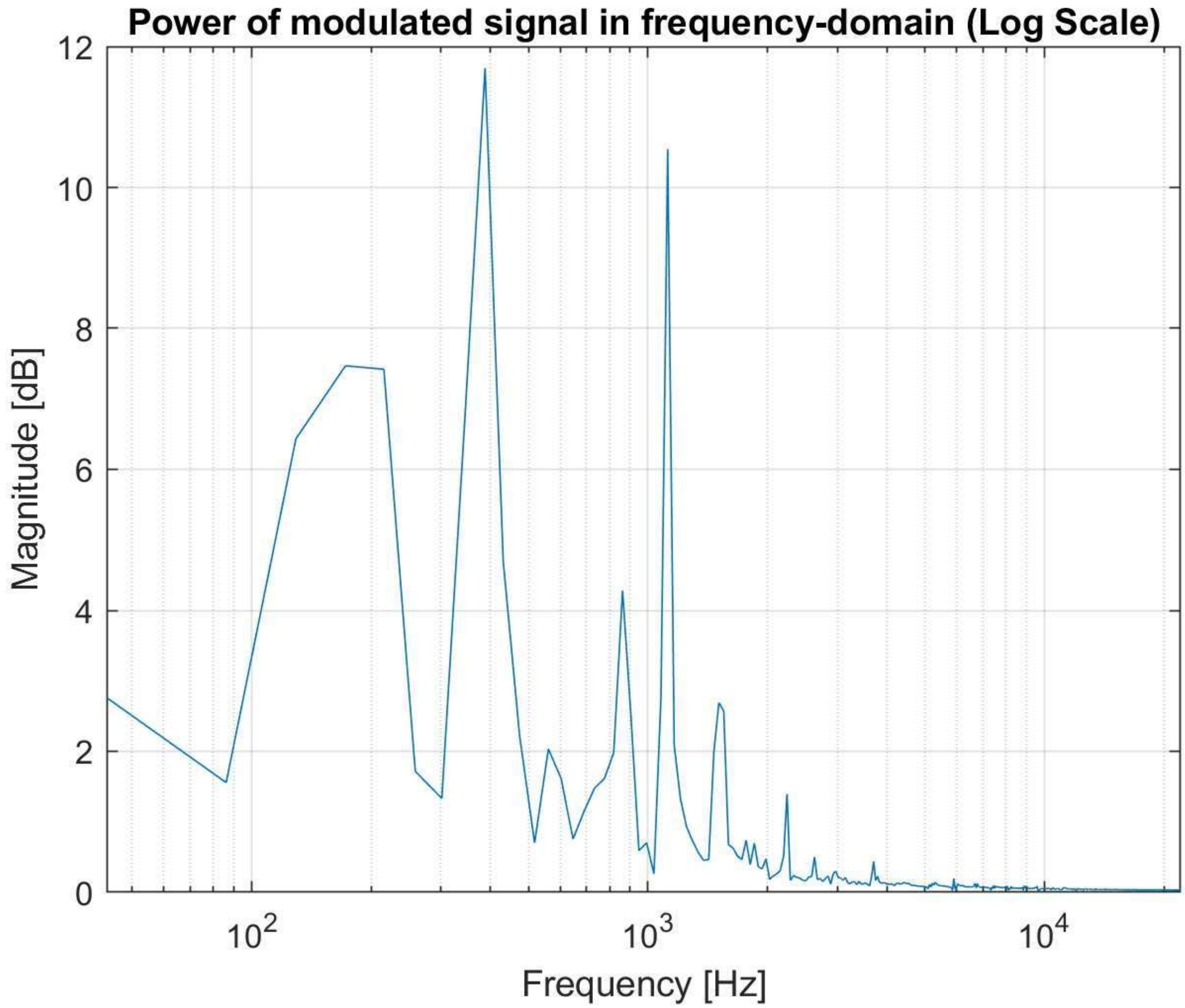
Shown below is a MATLAB plot of the output voltage in millivolts of the modulator across a 50Ω resistor:



Simulation Waveforms – Frequency Domain Comparison

Multiplying the 10GHz carrier signal by the modulating signal in MATLAB would require increasing the 44.1kHz audio sampling rate to at least 20GHz in order for the two signal vectors to have the same length. Unfortunately, this is too computationally intensive and causes the computer to overheat and freeze when trying to perform the upsamping process to allow multiplication between the two signals. Furthermore, the audio bandwidth is very small compared to the carrier frequency, therefore, calculating an FFT of the modulated voltage signal with enough frequency resolution to actually see the upper and lower sidebands would be far too computationally intensive. Therefore, we will settle with showing that the spectrum of the music signal is the same as the spectrum of the power contained in the modulated signal.




Power of modulated signal in frequency-domain (Log Scale)
Magnitude [dB]
Frequency [Hz]
0
2
4
6
8
10
12
10^2
10^3
10^4

```
% This script reads a music file and uses a clip of it to
% modulate the gain of a variable gain amplifier that amplifies
% the carrier signal

% File path and name
file_name = 'E:\tearstream-alley-262939.mp3';

% Attenuator and power amplifier attenuation and gain values in dB
attenuator = 15;
power_amplifier = 30;

% Carrier input power from oscillator in dBm
P_in = 0;
RL = 50;

% HMC694LP4 Gain Parameters in linear region
G0 = 15;
alpha = 50;
V0 = -1.225;
Vmax = -1.325;
Vmin = -1.125;

% read audio signal from the file and only take one channel of stereo
[stereo, fs] = audioread(file_name);
mt = stereo(:, 2);
mt = mt(fs*20:80*fs);    % Take a 60s clip

% Increase sample rate
% mt = upsample(mt, 10);
% fs = fs*10;

% define time axis
t = linspace(0, length(mt)/fs, length(mt));

% plot raw audio signal in time-domain
figure;
plot(t, mt);
title('Raw Music clip m(t) in time-domain')
xlabel('Time [s]');
ylabel('m(t) [V]');
xlim([0, 60]);

% Find the maximum value of the time-domain signal and normalize amplitude to VGA linear region
% and Bias message signal in linear region
max_value = max(mt);
mt = (mt / max_value) * 0.5*(Vmax - Vmin);
mt = mt + V0;

% plot the audio signal when bounded by the linear region of the HMC694LP4
figure;
plot(t, mt);
title('Raw Music clip m(t) in time-domain')
xlabel('Time [s]');
ylabel('m(t) [V]');
xlim([0, 60]);
```

```
% Gain of HMC694LP4 in its linear region
G = G0 + alpha*(mt - V0)

% plot modulated gain of the HMC694LP4
figure;
plot(t, G);
title('Gain of HMC694LP4 modulated by m(t)')
xlabel('Time [s]');
ylabel('Gain [dB]');
xlim([0, 60]);

% Output Power of attenuator and power amplifier
P_out = P_in + G - attenuator + power_amplifier;

% plot output power of the modulated signal in dBm
figure;
plot(t, P_out);
title('Output Power of Modulated Signal in dBm')
xlabel('Time [s]');
ylabel('Output Power [dBm]');
xlim([0, 60]);

% Output voltage envelope function of the modulator in millivolts
Vout_envelope = sqrt(2*RL*(10.^(0.1*(P_out-30))));

% plot envelope function of the modulated output voltage
figure;
plot(t, Vout_envelope);
title('Envelope of Modulated Output Voltage in millivolts')
xlabel('Time [s]');
ylabel('Output Voltage [mV]');
xlim([0, 60]);

% plot audio signal in frequency-domain
n = 1024;   % number of fft points
mfft = abs(fft(mt, n));

% define frequency axis
f = linspace(0, fs, n);

% Plot on logarithmic frequency scale
figure;
semilogx(f(1:n/2), mfft(1:n/2));
title('Music clip m(t) in frequency-domain (Log Scale)')
xlabel('Frequency [Hz]');
ylabel('Magnitude [dB]');

% Set the x-axis to a logarithmic scale and add grid lines
set(gca, 'XScale', 'log');

% Set x-axis limits to start from a small value (e.g., 10 Hz) and extend to the Nyquist frequency
xlim([fs/1024, fs/2]);

% Add grid lines for logarithmic scale
grid on;

% plot power of modulated signal in frequency-domain
```

```
n = 1024;   % number of fft points
mfft = abs(fft(P_out/30, n));

% define frequency axis
f = linspace(0, fs, n);

% Plot on logarithmic frequency scale
figure;
semilogx(f(1:n/2), mfft(1:n/2));
title('Power of modulated signal in frequency-domain (Log Scale)')
xlabel('Frequency [Hz]');
ylabel('Magnitude [dB]');

% Set the x-axis to a logarithmic scale and add grid lines
set(gca, 'XScale', 'log');

% Set x-axis limits to start from a small value (e.g., 10 Hz) and extend to the Nyquist frequency
xlim([fs/1024, fs/2]);

% Add grid lines for logarithmic scale
grid on;


% play the audio clipe
sound(mt,fs);
```

```
% This script converts the carrier frequency and power in dBm and converts
% it to a voltage signal through a 50 ohm resistor

% Define parameters
fc = 10e9;                  % 10 GHz carrier frequency
fs = 500e9;                 % 500GHz sampling rate
duration = 1*1e-9;          % 1 ns duration
power_dBm = 0;              % Signal power in dBm
RL = 50;                    % 50-ohm load resistor

% Carrier signal Amplitude in millivolts
A = sqrt(2*RL*(10^((power_dBm-30)/10)));

% Time vector
t = 0 : 1/fs : duration;  % Time steps based on sampling rate

% Generate the carrier signal
carrier = A*cos(2 * pi * fc * t);

% Plot the carrier signal
figure;
plot(t * 1e9, carrier);
title('10 GHz Carrier Signal in mV: 0dBm into 50-ohms');
xlabel('Time (ns)');
ylabel('Voltage [mV]');
grid on;
```