AM Demodulation

Topic	Student Name	Date	Course Leader
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This laboratory exercise expands the characteristics of the AM (amplitude modulation) linear demodulation. Usually an AM demodulator consists of a diode which has its parameters set to in order to perform the role of an envelope detector. In fact, this can demodulated using MATLAB Hilbert transform script; however this isn't the practice of this experiment as here we will be using a program called Tina TI.

Introduction

AM is a type of modulation that is very popular type of communication within electronic signals. An example of this could be a radio wave carrier wave. In order to introduce this further I believe that it'll be easier to introduce some theory from the lecture. For the sake of understanding let a(t) represent any arbitrary wave which is being transmitted (message) and let A represent its amplitude:

$$a(t) = A * \cos\left(f_c t + \emptyset\right)$$

Below please find the model of a sine carrier wave representation:

$$c(t) = A * \sin\left(f_c t + \emptyset\right)$$

For both the frequency [Hz] of an audio tone is given by:

$$w_c/2\pi \,;\; w_a/2\pi$$

Since we have all of the information provided already in front of us, we can now build y(t) being our amplitude modulation (AM) signal. And from this we know that our modulated signal is built up of three main components, a carrier wave and two sine waves. These sine waves are often referred to as sidebands.

$$y(t) = A * \sin(w_c t + \emptyset) + \frac{A^2}{2}\left(\sin(w_a + w_c)\, t + \emptyset\right) + \sin\left((w_c - w_a)t - \emptyset\right))$$

We can conclude this introduction section with the information that AM is different than frequency modulation and always must be taken into account as for frequency the carrier and phase signal is varied.

Measurements

All measurements for this experiment were simulated using the program TINA TI where we will assemble circuits for the AM signal generation. The question describing AM signal is:

$$s(t) = A_c \cos(w_c t + A_c k_{AM} m(t) \cos\left(w_c t\right)$$

As seen in the figure below, I have added two controlled sources as requested in the instruction manual. According to the given instructions and parameters I have set the expression as CS1 = V(N1)*V(N2) and number of voltages to two entries, later I proceeded to insert the generator with given VG1 and VG2 parameters.

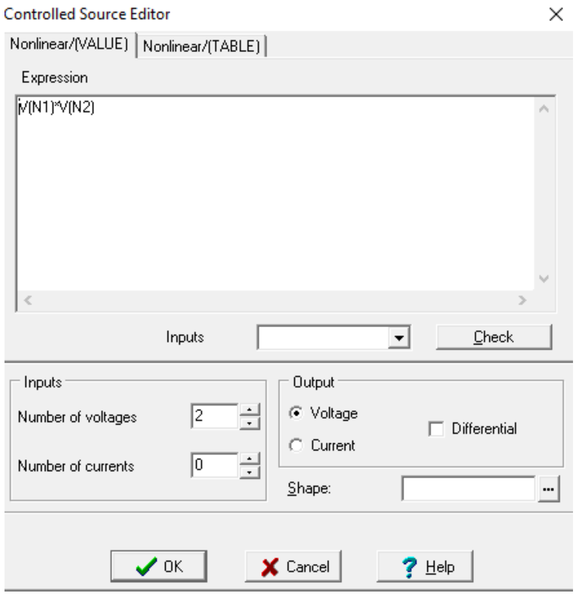

Figure 1.1 – Presents the process of specifying controlled sources.

Below please find the parameters that have been provided into the simulation for the voltage generators as given in instructions.

VG1 – Set to sine wave; amplitude of 5V; frequency 30 kHz (carrier)

VG2 – Set to sine wave; amplitude of 0.5V; frequency 1 kHz (message)

One is a multiplier controller CS1 whilst CS2 is the summing controller.

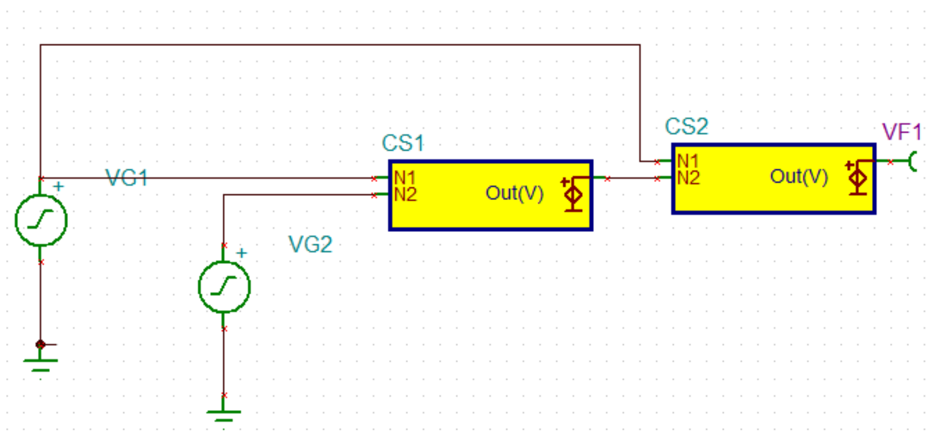

Figure 1.2 – Presents the generated voltage generator according to parameters provided. One controlled source and two voltage generators

In the below please notice an exported version of the graph drawn by our graph with use of the Transient analysis mode set to 2ms as stated in instructions. As we can see we have an output signal generated that has amplitude equivalent to estimate of 8V.

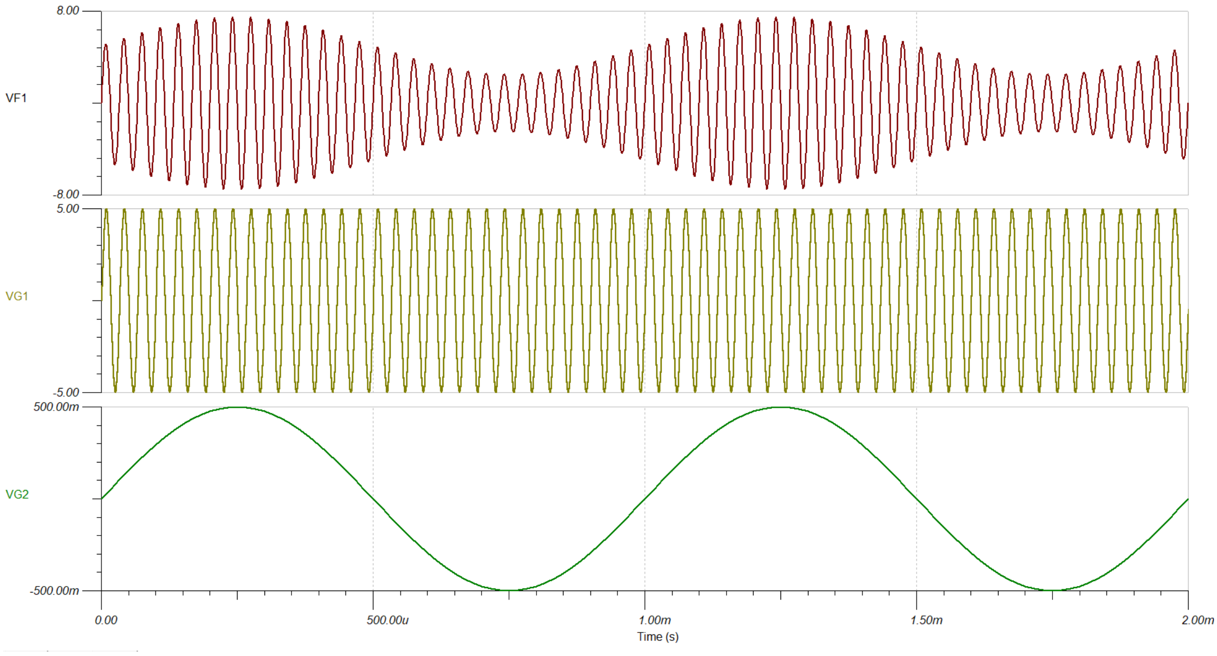

Figure 1.3 – Presents results of the transient state set according to provided parameters.

Maximum AM Amplitude: 7.5 V

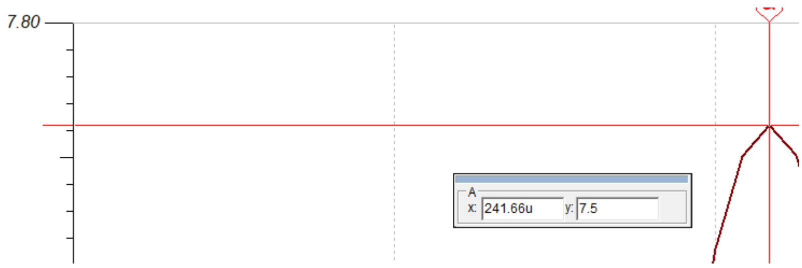

Figure 1.4 Presents the Maximal AM Amplitude (y axis value)

Minimum AM Amplitude: 2.5 V

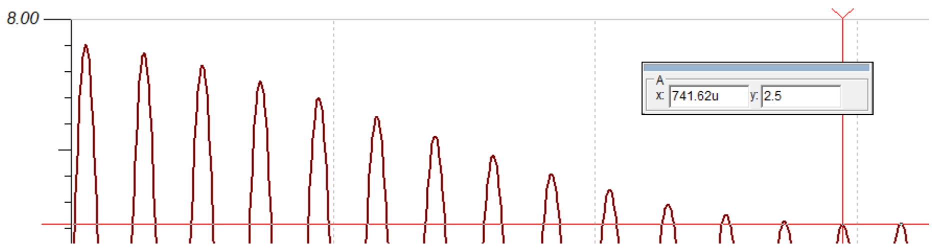

Figure 1.5 Presents the Minimal AM Amplitude (y axis value)

Now we will move onto calculating the AM modulation index. The modulation index is a measure of amplitude variation of a signal, usually about a demodulated or unmodulated carrier. My expectations are that if the modulation index is around 0.75 this would suggest the signal will increase by a factor of 0.75 and therefore decrease 0.25 of its original untouched level. Therefore a index of 1 would be the maximum level of modulation that can be applied to a signal since the envelope of a signal increases by 1… Expanding on this further, if we have TWICE of the steady value it will fall to zero simply speaking. Below is a provided equation that we will use in this laboratory to calculate our modulation index.

$$\mu = \frac{A_{MAX} - A_{MIN}}{A_{MAX} + A_{MIN}}$$

$$\mu = \frac{7.5_{MAX} - 2.5_{MIN}}{7.5_{MAX} + 2.5_{MIN}} = 0.5$$

From our observations and calculation, we see that our AM signal is around 0.5 modulation index, meaning that the signal will increase by a factor of HALF 0.5 sand therefore decrease by HALF 0.25 of its original signal.

In the below please find a figure that presents the new circuit with our new envelope detector. To build this I followed the instructions provided, where we used a diode, resistor and capacitor to build our circuit. Value of capacitor is 100nF for filtration, and resistor value is 1kΩ.

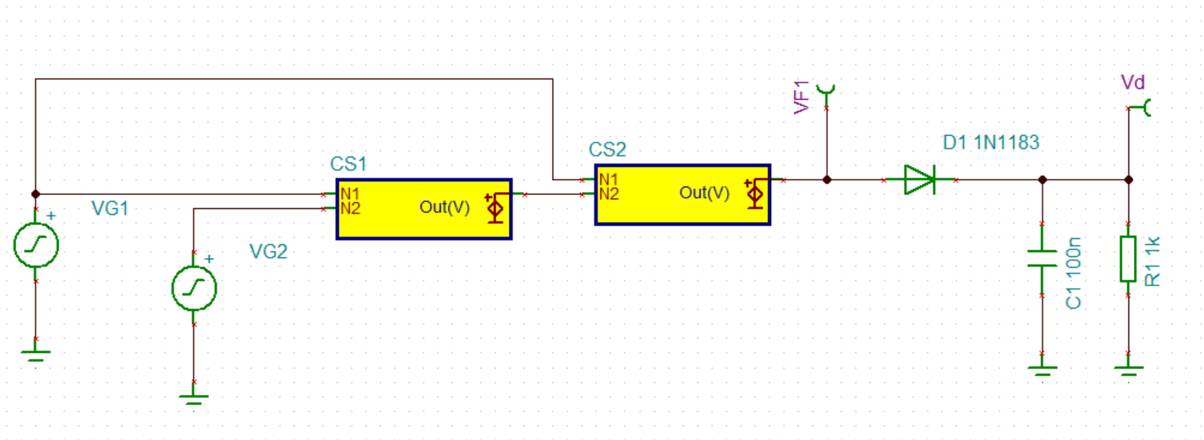

Figure 2.1 Presents new circuit with envelope detector using a diode, resistor and capacitor.

Below please see figure 2.2, which shows the new output simulation where can see the changes that were made to our signal in comparison to figure 1.3. We can see that the max output voltage of our signal is

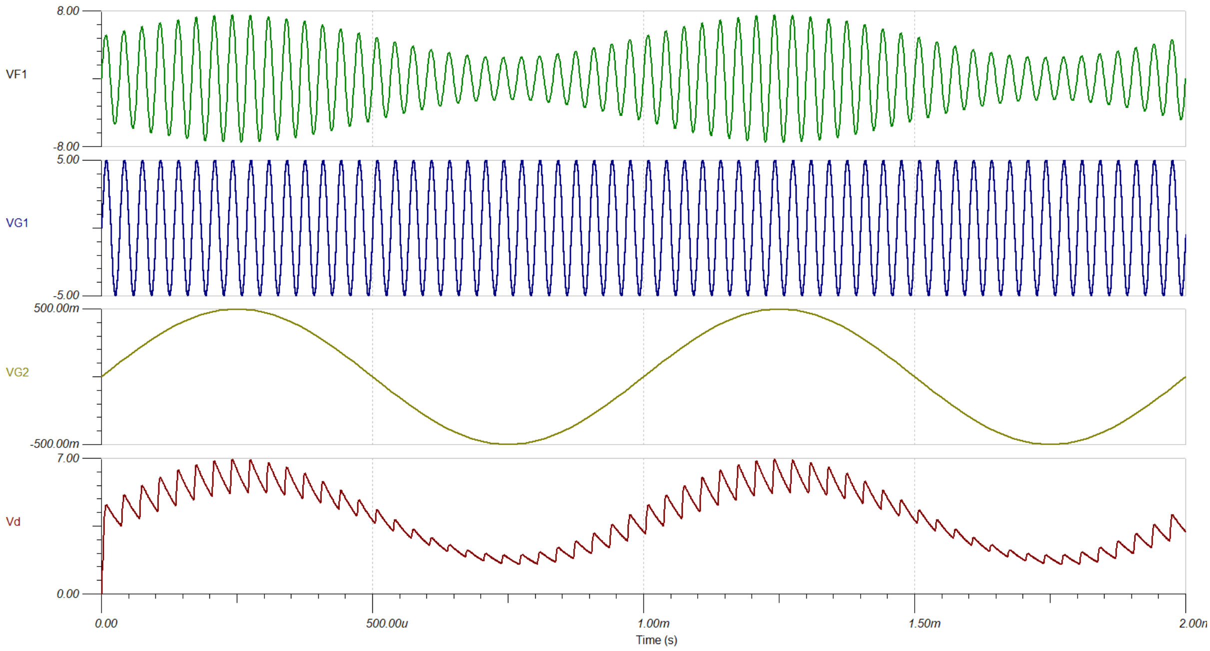

Figure 2.2 Presents simulation with envelope detector using a diode, resistor 1kΩ and capacitor100nF.

Below please find another transient simulation with changed parameters on the resistor R1 to 10kΩ

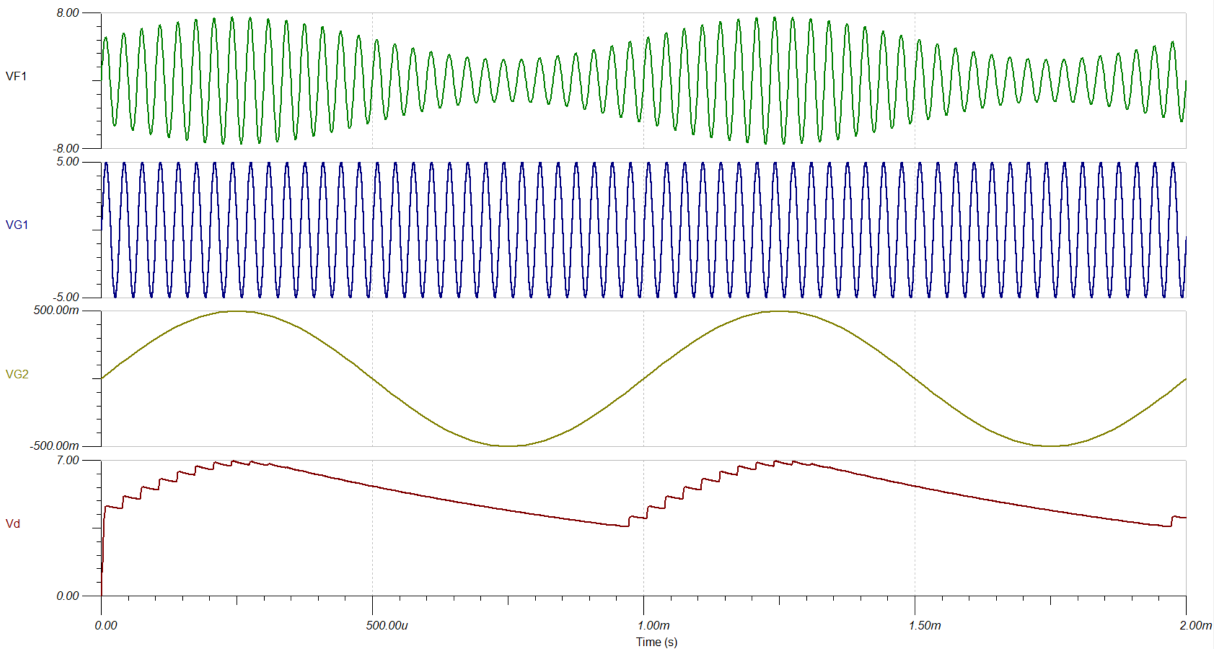

Figure 2.3 Presents simulation with envelope detector using a diode, resistor 10kΩ and capacitor 100nF.

Below please find another transient simulation with changed parameters on the resistor R1 to 100kΩ

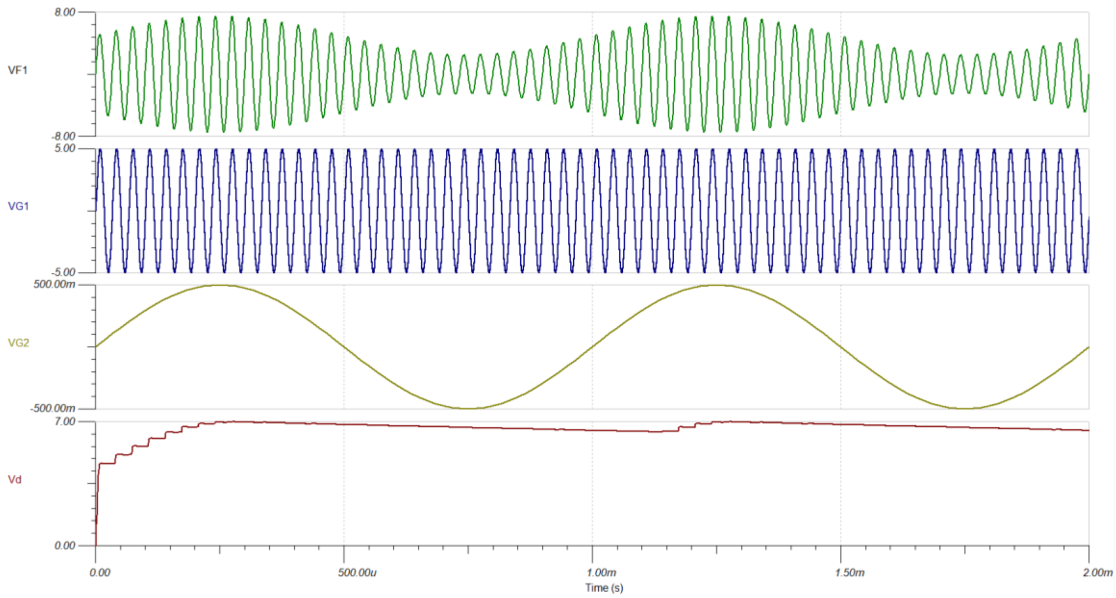

Figure 2.4 Presents simulation with envelope detector using a diode, resistor 100kΩ and capacitor 100nF.

The next part of our laboratory is to answer a question: How to assign a proper value of time constant in R1C1 filter? I believe that there can be various answers, initially my first thought is to gather data such as the initial steady-state voltage $V_{OUT}(0)$ and the final steady-state voltage $V_{OUT}(\infty)$ on the capacitor from, which we can calculate the output voltage using the equation provided below:

$$V_{OUT} = [V_{OUT}(0) - V_{OUT}(\infty)] * e^{-\frac{t}{T}} + V_{OUT}(\infty)$$

Then using this equation, we can more or less estimate out proper time constant for our filter. However, after many researches within book and trusted websites I came to an assumption that this is to be the most reasonable method.

Moving on forward please seek figure 3.1, which presents a newly drawn circuit.

Below please see figure 3.1, which shows a newly created circuit. This new circuit will now act as the 2nd order RC low-pass filter with the cutoff frequency of around 1kHz. All the values as well as the circuit design has been taken directly from the instruction manual.

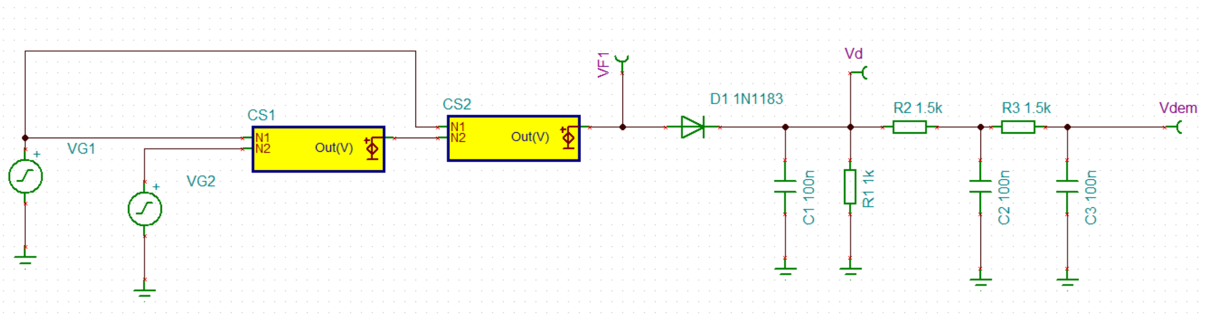

Figure 3.1 Presents new circuit acting as a 2nd order RC low-pass filter with cutoff frequency of around 1kHz.

***Remark**: Value of R1 has been reset back to its original setting of 1kΩ.*

Below please find figure 3.2, which presents the simulation of the circuit from figure 3.1

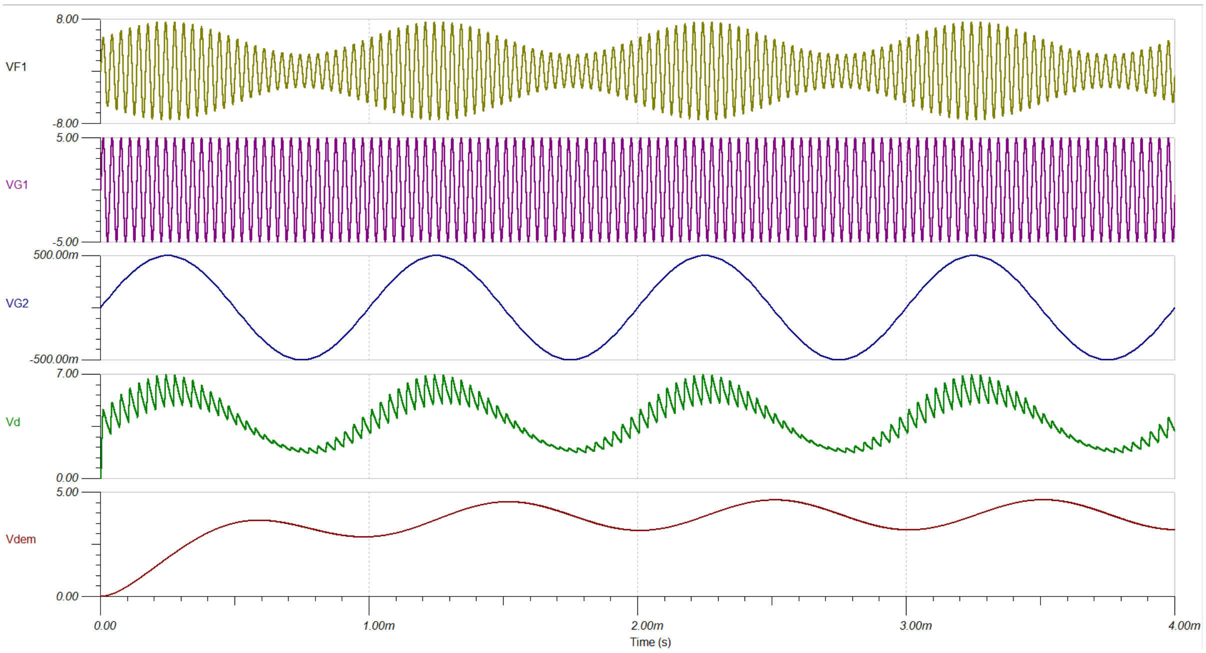

Figure 3.2 Presents simulation with a 2nd order RC low-pass filter with cutoff frequency of around 1kHz.

The next part of our laboratory is to answer a question: Why is necessary to use a low-pass filter on the output of the system? From the top of my head I can assume that the LPF will act as a smoothing filter, by doing so it ensures that the information we actually want to hear/have are being outputted – so technically speaking it ensures that the bandwidth of our interested is outputted. The reasoning supporting my answer is that in our circuit we are generating sinusoidal waves, these are then multiplied and then summed therefore, we generate multiples of the carrier frequency – these can be high frequencies in, which we have no interest therefore by applying a LPF we are able to remove them. Since on the output the signal is reconstructed by the LPF, depending on the depth of smoothing we have…here we have 1kHz cut off frequency, therefore we have successfully shown that we are able to adjust the smoothing depth. By smoothing depth, I am referring to our output response; depth refers to the decreased amount of high frequency components in our signal are not being passed through. Please refer to the figure below that shows an LPF "in action", I believe this will further ease the understanding and support my answer. Figure 3.3 is not referenced as it has been graphically created by me.

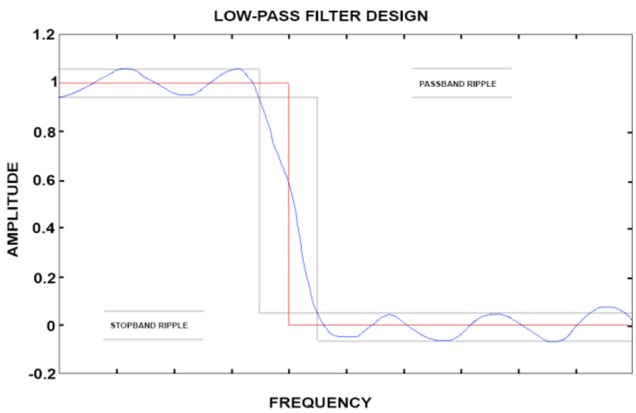

Figure 3.3 Presents an LPF "in action".

The next part of our laboratory is to answer a question: How to calculate the cutoff frequency of low pass RC filter? The answer of this question is as follows. All LPF have a certain rated cutoff frequency, meaning that it is a frequency above, which our output signal/voltage falls below around e.g. 70% of the input voltage however, please take into account that the provided percentage for the cutoff frequency isn't arbitrary. Since we are dealing with a RC LPF please find the equation below presenting how to calculate cutoff frequency.

$$f_{cutoff} = \frac{1}{2\pi RC}$$

As the instruction states VG2 signal has been set to the included .WAV format file and VG1 signal has been set to a sine wave with an amplitude of 2.5V and a frequency of 30kHz.

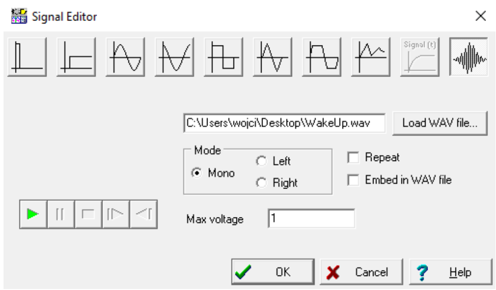

Figure 4.1 Presents new settings for VG2

In the figure 4 below please find a new circuit, the value of the cutoff frequency in the RC filter is around 3kHz. Please see that in the output of our circuit we happen to have a DC cutoff circuit.

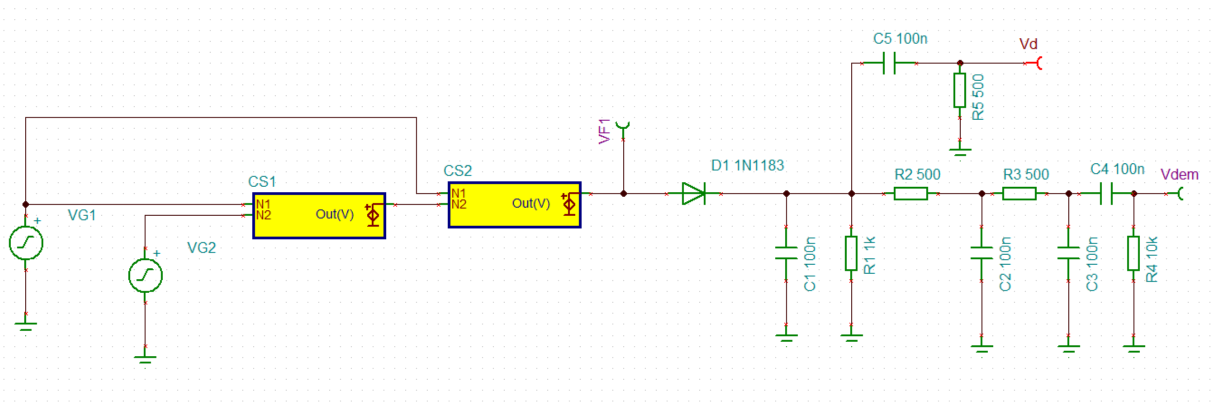

Figure 4.2 Presents new circuit acting as a 2nd order RC low-pass filter with cutoff frequency of around 3kHz.

Below please find figure 4.3, which presents the results of the simulated circuit from figure 4.2 (above). The sounds were played back and I have heard the VG1 signal being just a 30kHz signal, the original VG2 .wav signal, else at VF1 we can hear the multiplication of these signals as well as their summation and at the end of Vd : Vdem we hear the demodulation. Between Vd and Vdem we can hear a significant difference and some frequencies have been filtered out, I believe that this supports my answer to the question of why we need to apply an LPF at the end/output.

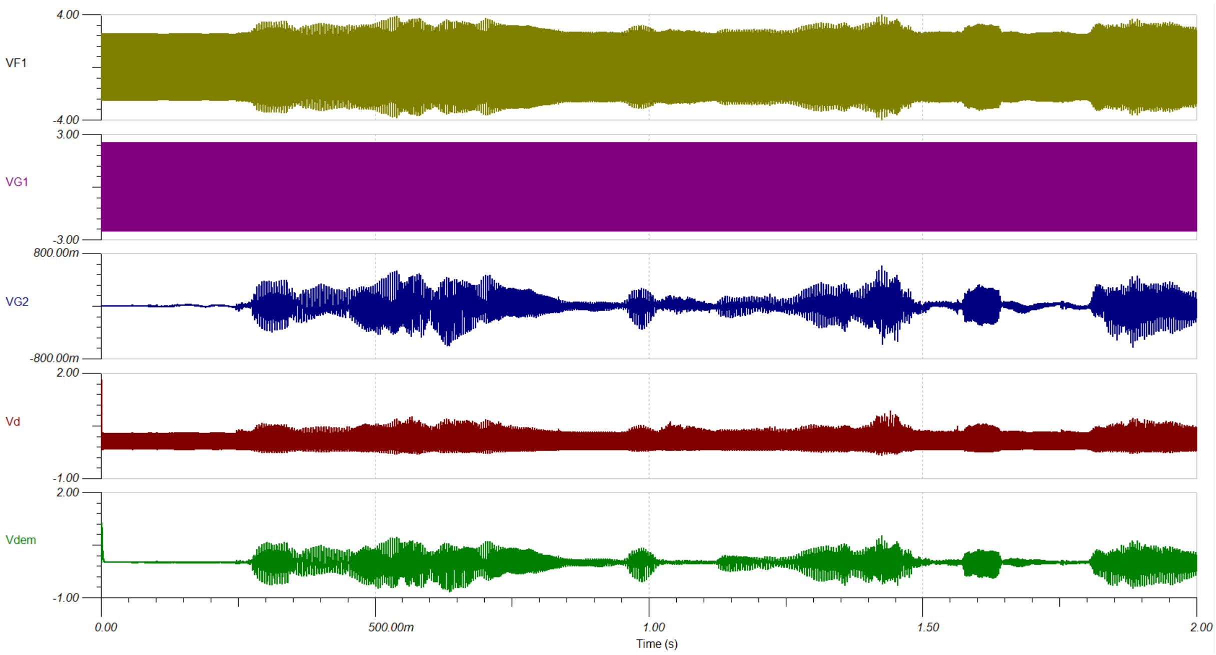

Figure 4.3 Presents simulation results from circuit presented in figure 4.2.

In the figure below, the values of R1 resistor have been changed to 10kΩ as per instruction provided in the manual.

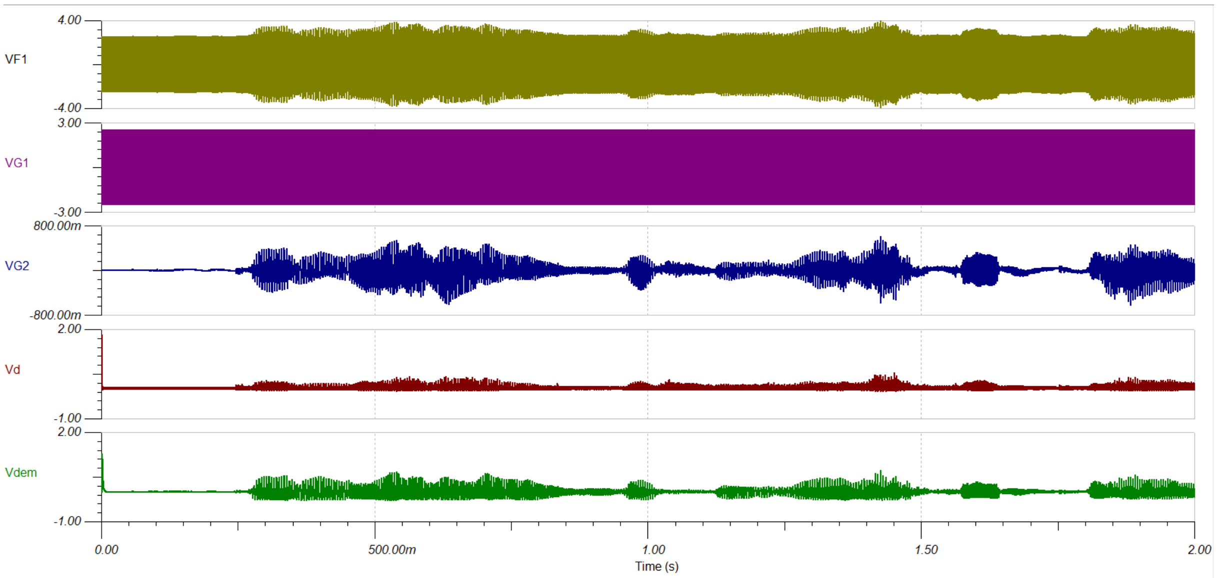

Figure 4.4 Presents simulation results from circuit presented in figure 4.2 with R1 value set to 10kΩ

In the figure below, the values of R1 resistor have been changed to 100Ω as per instruction provided in the manual.

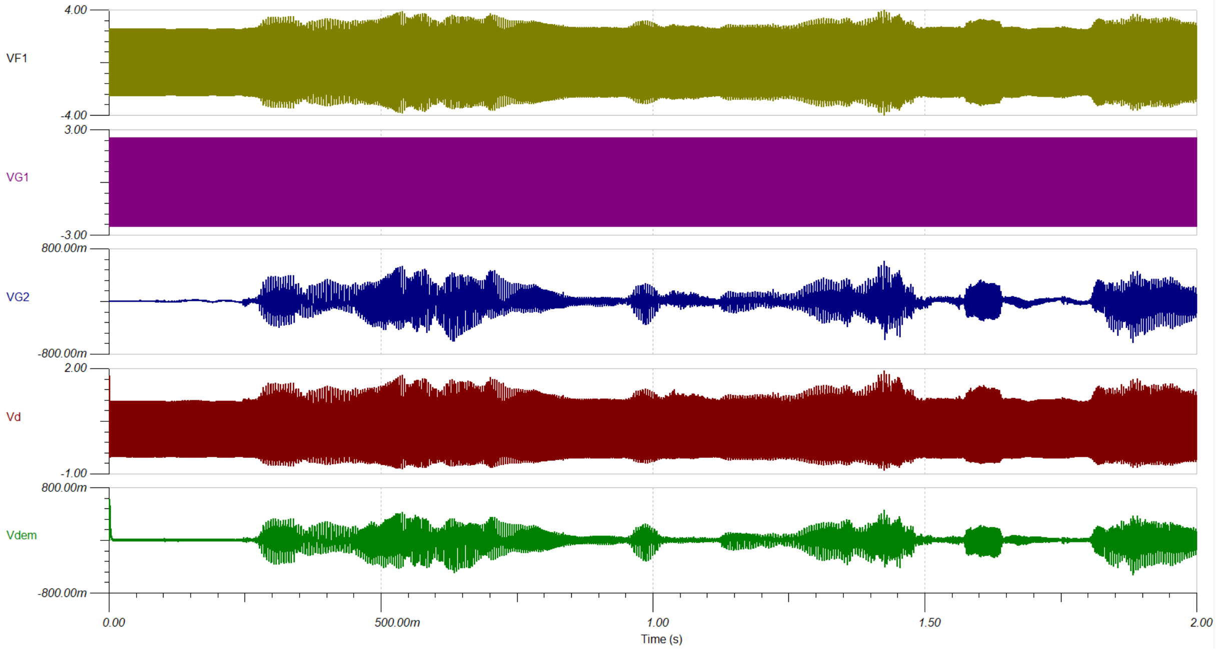

Figure 4.5 Presents simulation results from circuit presented in figure 4.2 with R1 value set to 10kΩ

The next part of our laboratory is to answer a question: How does the sound change and why? The sound changes, the lower the resistance the lower the level of filtration therefore we can state that by changing the value of R1 to a greater sum e.g. 10kΩ we manage to receive a much higher amount of filtration whereas the lower the sum the lesser the amount of filtration received. More technically, in our simple capacitive and resistive LPF we have a frequency that passes through which is known as the capacitive reactance and is usually provided in Ohms (Ω). The resistance affects the frequency response of our filter. The capacitor itself is a filter. However, we require some sort of impedance across a signal for the capacitor to filter, so impedance cannot be infinite or zero. In our case when dealing with lower resistive values for R1 we enable high frequencies to be passed to the output of the signal.

In the figure 5.1 below please find a new circuit, added is a new controlled source. The role of the circuit is to act as a synchronous AM detector. Using parameters provided in the manual, all voltage sources have been set to be generating a sinusoidal wave. VG1 amplitude of 5V with 30kHz carrier frequency; VG2 amplitude of 0.5V with 1kHz message frequency; VG3 amplitude of 1V with 30kHz heterodyne frequency.

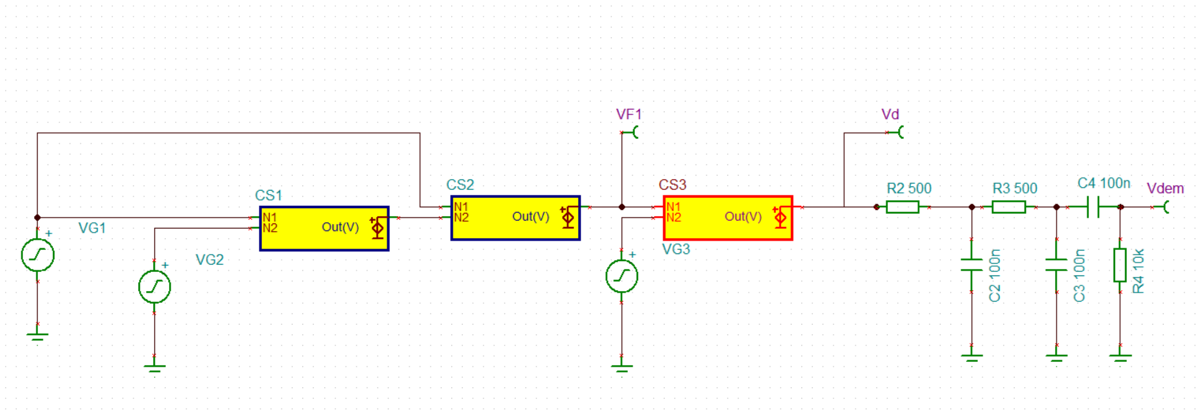

Figure 5.1 Presents new circuit acting as synchronous AM detector.

Below please find figure 5.2, which presents the simulation of the circuit from figure 5.1. (All plots have been magnified as results we're unreadable at normal 100% scaling).

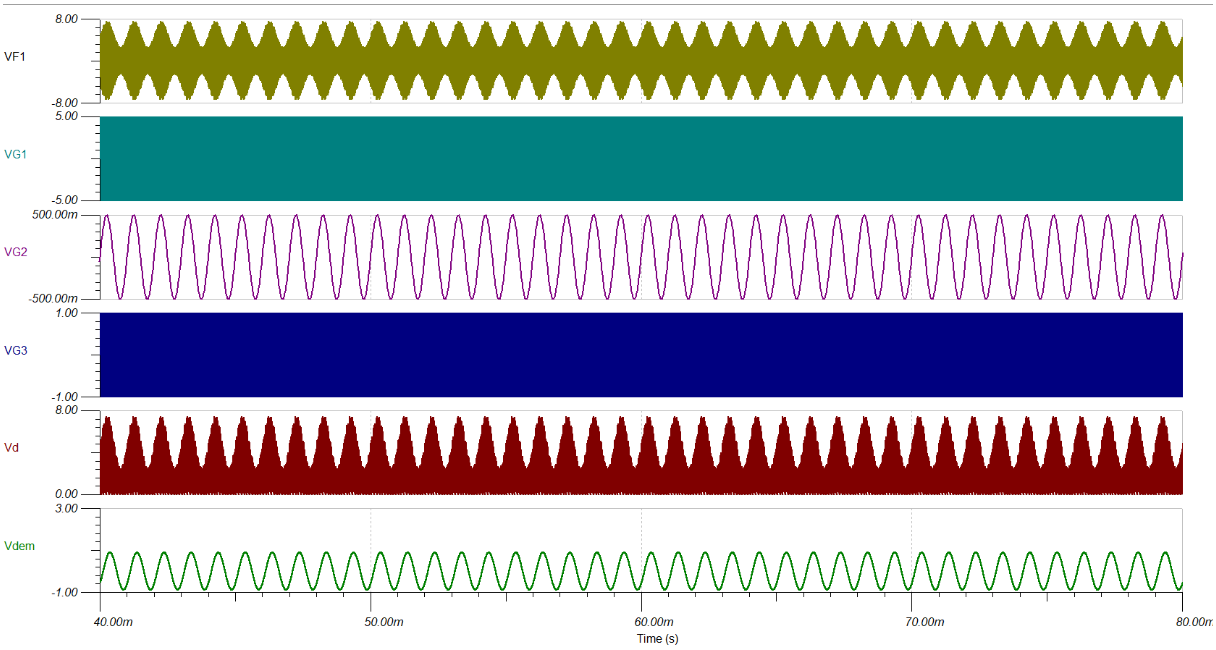

Figure 5.2 Presents simulation results from circuit presented in figure 5.1.

Below please find a figure presenting the same circuit with VG2 parameters changed to feed the .WAV file attached to instruction manual.

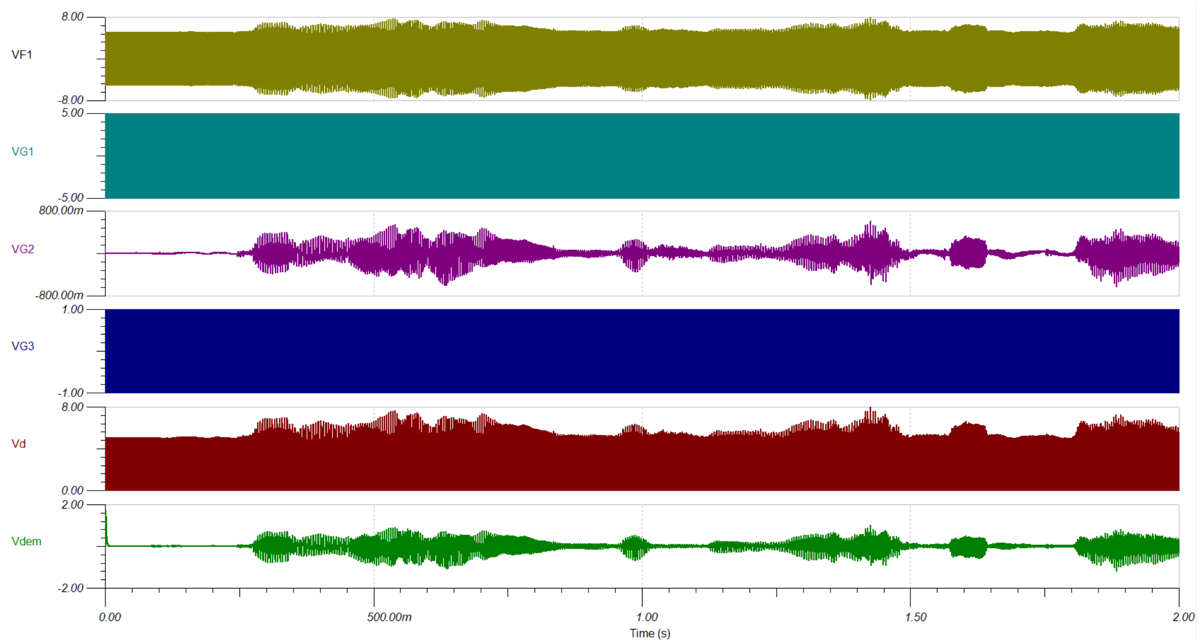

Figure 5.3 Presents simulation results from circuit presented in figure 5.1 with VG2 set to .WAV file.

I have changed the values of R2 and R3 to 1kΩ and as instruction said VG3 has been set to 30.02kHz exactly. From what the figure below shows we can observe that the output signal is fluctuating/oscillating in a sinusoidal manner therefore I can only assume that we have multiplied our signal past a certain point where we are able to actually impact the signal itself and maybe damaging it post demodulation – However, I'm not particularly sure with this statement therefore please only treat this as my assumption.

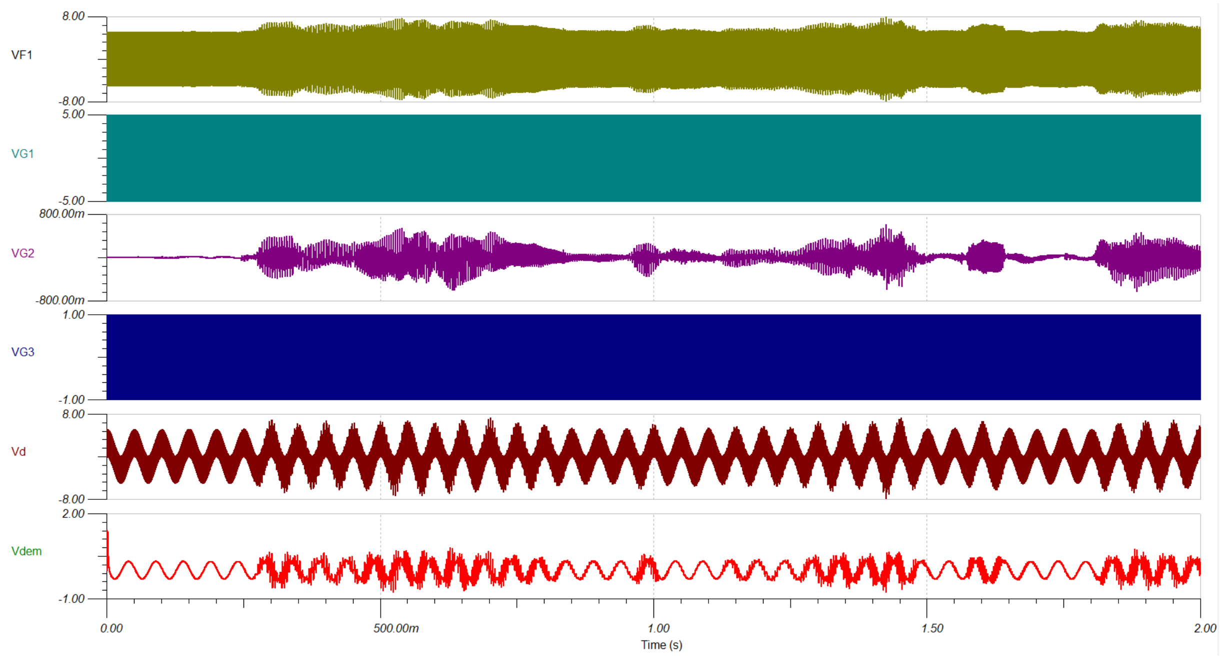

Figure 5.4 Presents simulation results from circuit presented in figure 5.1 with changed params.