Introduction to Signal Processing Using MATLAB and Simulink

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Introduction

As a second year Engineering student, you have been introduced to a variety of tools to handle the theoretical aspect of science (using mathematics, physics, and algebra) and its applied side (using programming languages, circuits and some design). The latter, not surprisingly, is primarily what Engineers do. Yes, Engineers with capital E.

In the past weeks you have been introduced in ECE221 (Electricity and Magnetism) to `Matlab`, a numerical simulation package. At this point you have already observed that `Matlab` looks and feels like a programming language. It comes across as a tool that sits on the border between the theoretical and the applied sciences. You can use it to plot a variety of theoretical curves, or change variables in equations to observe outcomes dynamically. As you will see in the future, you will be able to create "inputs" to equations in the form of curves, and plot "outputs", as if the behaviour of a system under test were described by your equation.

Your introduction to `Matlab` was in the form of individual commands entered at the prompt, functions written and called from the command line, or full programs run within `Matlab`. In this experiment you will expand your `Matlab` experience further. First, by making `Matlab` interact with the outside world (i.e., with the file system and the speaker/headphone output on your workstation), and by using `Simulink`.

If you saw `Matlab` as a numerical computation environment, `Simulink` is its visual counterpart, in which you have control of when things start and stop, and you can define inputs, observe outputs and determine the behaviour of various systems

under test as a simulation model runs. You can think of the "running" as inputs changing along time and producing outputs according to a certain function. This will become clearer as you do the steps in this experiment.

At the end of this experiment, it is expected that you will be very familiar with:

- Making `Matlab` talk to the outside world;
- Building simple simulation models in `Simulink`;
- Generating and displaying simulated signals;
- Making `Simulink` interact with the outside world.

As you work through this lab, there will be exercises for you to answer. Parts of an exercise are numbered (a),(b),(c), etc. For example, Exercise 1.2 (c) is part c of Exercise 1.2 in Section 1.2. Some exercises may only have part a.

1. Matlab

Start Matlab using any of the methods that have been provided to you. Matlab is also available online. If you wish to do the lab using Matlab Online, it is best to do it in a supported browser like Chrome as not all browsers are compatible with Matlab's sound capabilities.

1.1 Creating and Plotting a Sinusoid

In this section you will write a program to generate a $1KHz$ sinusoid and play it to the outside world (that is, your speakers/headphones). You can write your program on the `Matlab` text editor or can write individual lines at the command prompt. It is better, for editing, if you write in a program. If you do write as a program, don't forget to use `.m` as the extension when naming your program file.

The program below generates 1 second of a 1KHz sinusoid using a sampling rate of 48000 samples per second. In other words, the output of this program is a vector with 48000 points, which represent a 1KHz sine wave.

```
T=1;
Fs = 48000;
N = T*Fs;
t = 0 : 1/Fs : T;
Fn = 1000;
y = sin(Fn*2*pi*t);
plot(t,y);
axis([0 48/48000 -1 1])
```

You can think of the output of this program as a sequence of values that follow a sinusoidal pattern. The values are spaced evenly, according to a time interval that you set. You can think of an *analog* signal as the one where the this time interval is infinitely small inbetween samples. In *digital* signals, each of these pieces – or *samples* – of the sinusoid can be identified, as they happen at a specified time interval and are given a fixed value. In other words, they are taken "frequently" at a certain sampling rate (or sampling frequency), and are assigned a value following a sinusoidal function for this particular case.

Exercise 1.1:

(a) After having understood what the program does, your task is now to modify the program to plot three periods of your 1KHz sine wave. Show the plot to your TA.

1.2 Listening to a Sine Wave

In the program above, you generated a 1KHz sine, but have you ever heard one? This is what you will do now. First, before everything, bring the volume on your workstation down to a low level (close to zero). This is accomplished using the audio controls. Now plug the headphones of your smartphone/mp3 player into the headphone jack of the workstation. When you change volume levels, the computer sounds an audible bell/chime as a reference to you. Make sure you can hear this chime when you change the audio level, and – most importantly – *be sure that the level is comfortable to your hearing*. Keep the level to as low as possible, but with the chime still audible. You can keep the transducer (i.e., the earbud) at a short distance from your ear and still hear the signal.

The program below is almost exactly the same as the previous one, but the last line `sound(y,Fs)` sends the sinusoid to the earphones.

```
T=1;
Fs = 48000;
N = T*Fs;
t = 0 : 1/Fs : T;
Fn = 1000;
y = sin(Fn*2*pi*t);
sound(y,Fs);
```

Exercise 1.2:

(a) Play the program and hear the 1KHz sine wave;

(b) Change the frequency to 500Hz and play it again;

(c) Now change it to hear 2KHz and play it again.

(d) Now change back to 10KHz and explain what you have observed/heard.

Before you perform the next exercise, it will be more comfortable if your frequency is set to 1KHz.

- Your task now is to double the volume of your sinusoid modifying one line of your code. Again, set your audio level in the computer to the bare minimum, modify and run your code. As a precaution, keep the transducer (the earbud) at a short distance from your ear, so that you will be able to hear the difference without hurting your hearing.

Your transducer is a coil (an inductor) that causes a membrane to vibrate proportionally to the change in voltage applied to the coil. To the output stage of the audio card in the computer, the transducer (or the coil) is a load. As you double the voltage, the membrane moves farther (*not more frequently!*), giving you the perception of a higher volume.

(e) As you doubled the voltage, what is the change in dB of the signal as measured at the load? Note that the dB you are calculating here *does not* represent sound pressure; you are only comparing voltages.

1.3 Audio Signal Processing–A Musical Preview

In this section you will get your feet wet by looking at how you can use Matlab to process real-world signals. Real-world signals are considered continuous-time signals, and in order to be processed digitally and be stored on the computer, they must be sampled and quantized. How fast the signal is sampled is called the sampling rate (or sampling frequency), and how many bits per sample is determined by number of bits in the quantization.

1. First, download three ".wav" files from Blackboard to your current working folder. They are "bass.wav", "drums.wav", and "guitar.wav".

2. Next, you need to feed these real-world sounds into Matlab. In order to do so, there is a very useful function called "audioread" that takes in a ".wav" file and outputs the sampled data of the audio file, as follows:

   ```
   [bass, fs, nbits] = audioread('bass.wav');
   guitar = audioread('guitar.wav');
   drums = audioread('drums.wav');
   ```

   Each of the "bass", "guitar", and "drums" variables store a vector of sampled values of the corresponding continuous-time signals. If you look at the lengths of each of these sampled vectors (which indicates the number of sampled values), the numbers are huge! DO NOT display these sample values in the command window as it will take a long time to display all of them and therefore will likely freeze the screen. If you accidentally did this, press `CTRL-C` to stop the display (or wait for it to finish).

Exercise 1.3:

(a) Examine the length of the guitar signal (via: `length(guitar)`). The sampling rate is stored in the variable `fs`, and is in units of "samples per second". The value of "fs" is the same for all three sound signals. Using these facts, determine, in seconds, the duration of the guitar signal.

3. Copy and paste the following code into the editor or command window to create three sounds lasting 5 seconds each.

```
duration = 5;
b = bass(1:fs*duration);
g = guitar(1:fs*duration);
d = drums(1:fs*duration);
```

4. The Matlab function ”sound” plays back the sampled audio signal. To use ”sound”, you need to feed it the audio data as well as specify a sampling rate, in this case, ”fs”.

   ```
   sound(g, fs);
   ```

   Now try to hear the ”b” and ”d” sound signals in a similar way.

(b) Here comes the punchline: you can synthesize sounds by combining sound data digitally, as follows:

```
comp = b + g + d;
sound(comp, fs);
```

Play it for the TA to hear.

(c) Now write a function or script that takes in ”b”, ”g”, and ”d” audio data and outputs a five-second audio clip that includes all three of these instruments, but this time gradually increase the volume of the guitar sound. Show the audio clip and completed code to TA. (Hint: consider processing the guitar signal with a ”ramp” function, and linspace is a useful function here). Caution: signal amplitude gain should not exceed 5, or you will hurt your ears!

2. Simulink

You have now seen enough of Matlab to be able to use it as a tool in your studies. Now you will concentrate on Simulink, which is the visual simulation package that complements Matlab. Matlab and Simulink are complementary tools, and you can apply the same understanding and terminology you have from Matlab to Simulink.

When making Matlab interact with Simulink, the command window is still the window in which commands are written and parameters are set. If you set a parameter in a Simulink block, it will be visible to Matlab. When you change a parameter in Matlab and it is being used by a block in Simulink, it will be changed globally, and affect your block in Simulink. Keep this in mind.

The Simulink Library Browser icon is presented in Figure 1, and is found among the icons at the top of the main Matlab window you have opened. The icon is almost in the middle of the menu. Double-click on it now.

Figure 1. The Simulink Library Browser Icon

This will open the Simulink library browser window, as presented in Figure 2. This is where you will open new and existing simulation models and where you will find all the blocks you will need throughout this experiment.

Upon inspecting this window, you will find not only the basic Simulink blocks, but also all the toolboxes installed in your workstation. For now, take some time and explore the subsets within the main Simulink block set. You can do that by clicking on their name and inspecting the blocks. You can double-click on the blocks to see the parameters, but since the block does not yet belong to a model, you will not be able to edit the parameters, just view them. Since you are in your second year of studies, you should be somewhat familiar with some of the blocks in the blocksets labeled Continuous, Logic and Bit Operations, Math Operations, Signal Attributes, Signal Routing, Sinks and Sources. This is to say that you have heard of many of these *names*, but perhaps never combined various of these parts into a large, more complex, system.

You may have also noticed that in Figure 2 there is one item circled. This is the icon you select to open a new model in Simulink. This is what you will start next.

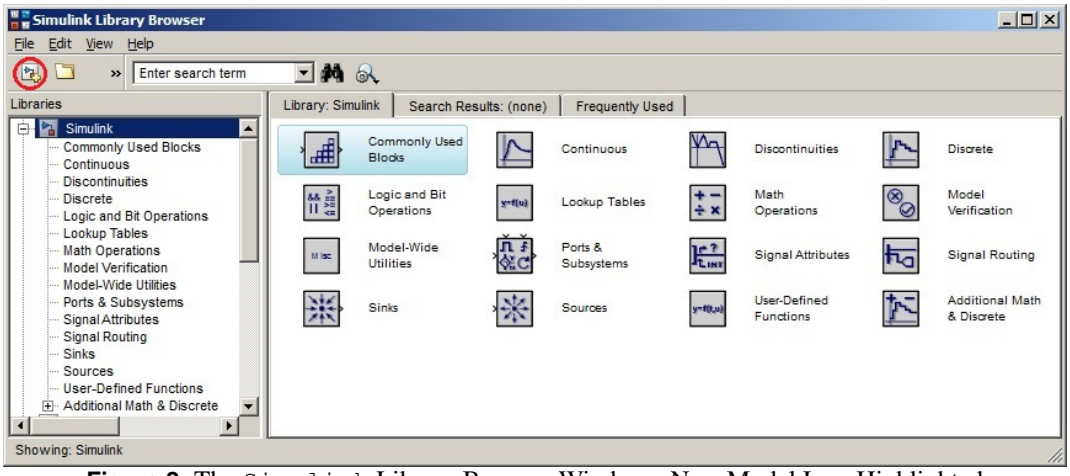


Figure 2. The `Simulink` Library Browser Window - New Model Icon Highlighted

2.1 First Simulink Model

From the `Simulink` library browser window, open a new model. You can do this in different ways: 1) by typing `Ctrl+N`; 2) by going under `File -- New -- Model`; or 3) by clicking on the icon highlighted in Figure 2. This will open an empty model. You will now click on desired blocks from the library browser you opened before and drag them to your empty model.

If you want to simulate a sine wave, you bring two blocks into your new model: the *source* called `Sine Wave`, and the *sink* called `Scope`. When the two of them are in the model, you connect the output of the `Sine Wave` to the input of the `Scope`. That is, you click on the output port and drag the arrow to the input port of the next block and let go. The output of one block goes into the input of the next block, just like in the lab. Now save the model and name it as you wish. When you are done, it should look like Figure 3 below.

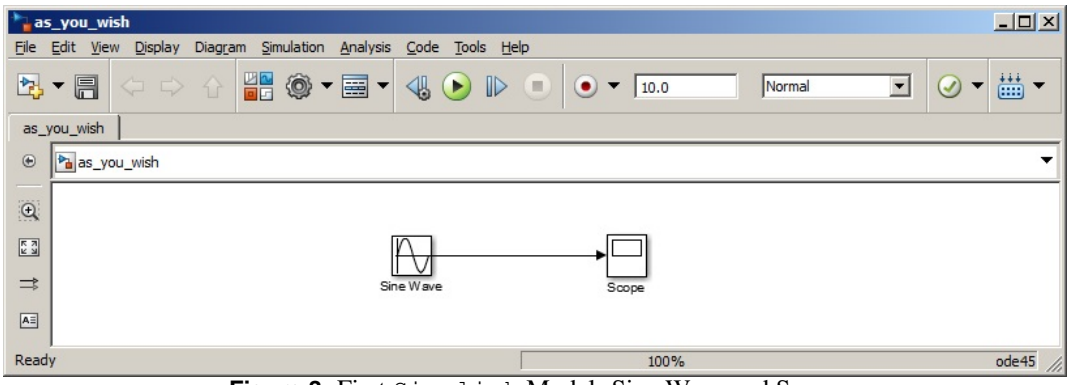


Figure 3. First `Simulink` Model: Sine Wave and Scope

We now want to simulate the system we have built. When we *simulate*, we are making things run along a time base. This time base is dictated by `Simulink`, but it is meant to represent "real" time, or seconds. The top menu of your model presents a space where you can determine your desired simulation time in seconds, and when you run the model it displays the simulated time in seconds as well. If you time it on your watch, they may not match, but as far as your signals are concerned, all the time labels are in seconds (just like on an oscilloscope).

In 3 above, if you double-click on the sine wave block, you can define the frequency of your sine wave, in radians per second. So if you set your frequency to be `2 * pi * 2`, and your simulation time to 1 sec, you should see two periods of a sine wave displayed on the oscilloscope.

Another way to set the parameters for the sine wave is to use the *sample base* type of simulation. When you use this type, you must set your frequency in samples per period, and you must define a sampling time. The latter is the interval between samples of the sine wave. For instance, mp3 songs ripped from CDs typically use a sampling rate of 44100 samples per second, which translates to a sampling *frequency* of 44.1KHz. The mp3 format allows for a maximum of 48000 samples per second.

Let's try it now. The objective is to display a 1KHz sine wave at a sampling rate for high audio quality. Set your sampling time to `1/48000` to start (that is the "high audio quality"). Now you want to display a 1KHz sine, so how many samples are there per period? As you made these changes, note that the icon for your `Sine Wave` block changed automatically to show

that you are now using samples. After you have set everything up, run the simulation for 1 second. Everything looks perfect... until you double click on the scope!

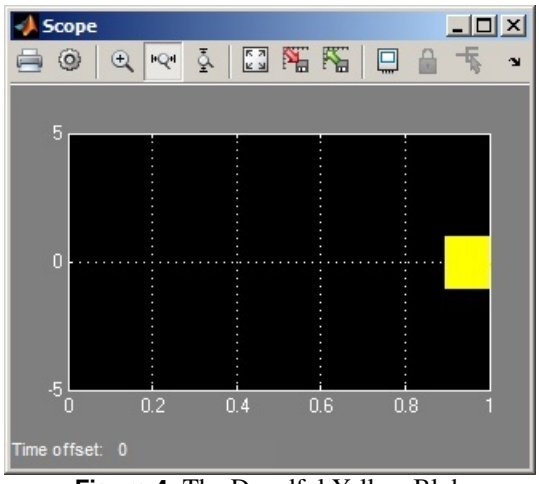


Figure 4. The Dreadful Yellow Blob

What you likely see when you click on the scope is the Dreadful Yellow Blob. How can you tell that this is, in fact, your 1KHz sine wave? You need to use the zooming function of the oscilloscope to achieve a meaningful display. The default setting is more confusing than helpful. Hover the mouse over the icons at the top of the scope display. You will find four tempting options: *zoom*, *zoom-X*, *zoom-Y* and *autoscale*. Resist your temptation to go for the *zoom* and *autoscale*. Instead, use *zoom-X* and *zoom-Y* to display your sine wave so that it does look like a sine wave, and then press on **save current axes settings**. The last step is key; if you do not save your axis settings, the display will return to the Dreadful Yellow Blob the next time you run your simulation. So now that you have saved your axes settings, run your simulation and see what happens.

Exercise 2.1:

(a) When you are happy with your simulation, show your 1 KHz sine wave with $\frac{1}{48000}$ sampling to the TA.

This *sample base* type of signal is the one you will use to interface with the "outside world" during your simulations. This means collecting a signal from a file or from a microphone, or sending a signal to headphones. You have already done something similar from the command line using `Matlab` and you will do some more of that in this experiment.

It is worth keeping in mind that the amplitude of the signals presented in the simulation is meant to represent voltages, or differences in electrical potential relative to ground. For the simulation, it is reasonable to assume that ground (or the reference) is the origin, or 0 Volts. `Simulink` does not say it explicitly, but if your Y axis ranges from -1 to +1, you can think of the unit as Volts, as if it were "the real thing". If the simulation were not the same as "the real thing", there would not be a point to running the simulation. When you did the `Matlab` part of the lab, we discussed assigning a 16 bit number to every sample [1].

Now that you know almost everything about generating a signal, let us do some operations on signals.

2.2 The Four Operations (`+ - * /`)

If you think of your sine wave as a sequence of individual points equally spaced along time (that would be your X axis) and with values representing Volts (that would be your Y axis), it should be easy to think of operating mathematically on this sequence. When making reference to sinusoids, Electrical Engineers refer to *frequency* and *amplitude*. Frequency is specified in Hertz (Hz), and amplitude in Volts (V), either "Volts-peak" (V_p) or "Volts-peak-to-peak"(V_{pp}). The latter refers to the difference between two crests of the signal, whereas the former refers to the difference between the ground – which is your reference – and the (highest) positive crest.

2.2.1 Adding or Subtracting a Constant to Sinusoid

Now that you have already built a model with two blocks, you can easily bring in some more blocks to make your model more complex. Drag in two more blocks: `Constant` and `Sum`. Connect them as shown in Figure 5. You may have to delete the existing connection before placing and connecting all blocks (there are different ways of doing it).

Now run the model and view the output on the scope. Figure 6 shows a side-by-side comparison of the output before, and after, adding a constant to your sine. In Electrical Engineering, this is called a *DC shift*. That is to say, you have a signal of a

[1] In the process of extracting samples from a signal, this step is called *quantization*, or "assigning a number value to the amplitude".

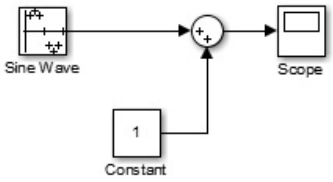
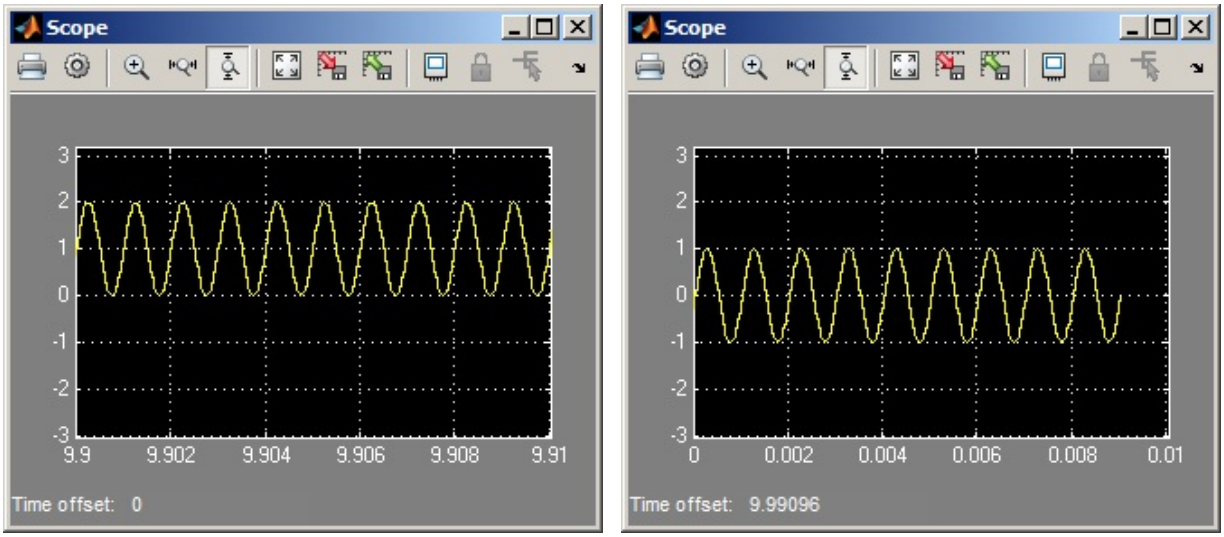


Figure 5. DC Shift

certain level (voltage) varying in time and you add a constant voltage (like a battery) to it. This happens, for instance, when you have a microphone that needs *biasing* to work. A DC voltage provides the microphone with the potential it needs to work, and the voice it picks up (which will be converted to voltage that varies in time) is added to this DC voltage prior to going into the amplifier. The amplifier will have a capacitor right at its input to block the DC and let the voice pass on to be amplified.



(a) DC Shift (b) No DC Shift

Figure 6. Two Sinusoids: Without DC Offset and With DC Offset

Exercise 2.2.1:

(a) Modify the constant to show the TA a DC-shift of 2 for the sinusoid.

2.2.2 Gain (call it "Multiplication by a Scalar")

A gain is a multiplication by a factor, say β. If β is less than one and greater than zero, you have *attenuated* the input signal (really a division!). If β is negative, then the signal is also "flipped" across the Y axis, which really means a phase shift by 180 degrees.

Figure 7. 6dB Gain

After you add the gain block to your model, try different factors and observe the result. Why is Figure 7 labeled a "6dB Gain"?

Exercise 2.2.2:

(a) Apply a 10dB gain to your sine wave and show it to the TA.

2.2.3 Operating on Two Sinusoids, Making Them Sound via Speaker

Build the model to have two sine wave inputs: one of them at 1KHz and the other at 2KHz. Now apply a gain of 2 to the 1KHz sinusoidal tone, add them and plot. Figure 8 is what your model should look like. Run your model and observe the output.

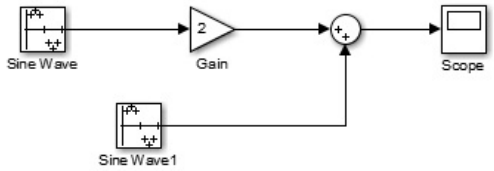


Figure 8. Two Sinusoids, one with a gain

Now you will listen to the sounds that your simulation generates, and will change parameters while the simulation is running. In order for you to do this, first you will need to send your signal to the standard audio device of your workstation. Go to the `DSP System Toolbox -- Sinks`, and drag into your model the `To Audio Device` block. Link it to the same signal path as your `Scope`. You can give this new model a run as it is, but it can be improved with a variable gain, as you will see below. In case you make it run now, be sure to put the headphones/earphones at a distance from your ear, until you find the right level to use.

What you need to do now, is to substitute the `Gain` block associated with the gain of 2 by the `Slider Gain` block. Also, add a `Slider Gain` to the other path as well. This block is found in the library `Simulink -- Math Operations`. You need to define the range of gains to be applied to the signal on each path. Suggestions are from 0 to 2 on the fundamental signal (your 1KHz sine) and from 0 to 1 on your 2nd harmonic (your 2KHz sine). You final model should look like Figure 9 below.

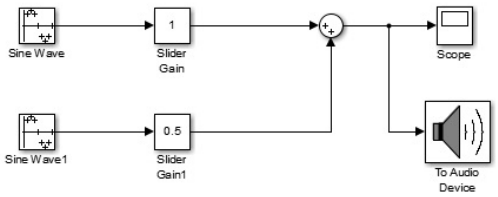


Figure 9. Two Sinusoids, Two Variable Gains, Sounding via Audio Device

With this model, let us simulate harmonic distortion. Musicians often refer to the addition of harmonics onto the original signal as "coloration". Harmonics can be added in a variety of ways. Some are *wanted*, like sound deliberately put through overdriven tube amplifiers, and some are unwanted, like the 2nd harmonic "hum" added by a transformer in the rectifier/power supply stage of an amplifier. Now run your model and change slowly the gain on the 2nd harmonic and see what audible difference it makes to the signal. Is it "more colourful"? When you run your model, make sure to have the `Scope` display the waveform, so that you can correlate what you hear with what you see.

Exercise 2.2.3:

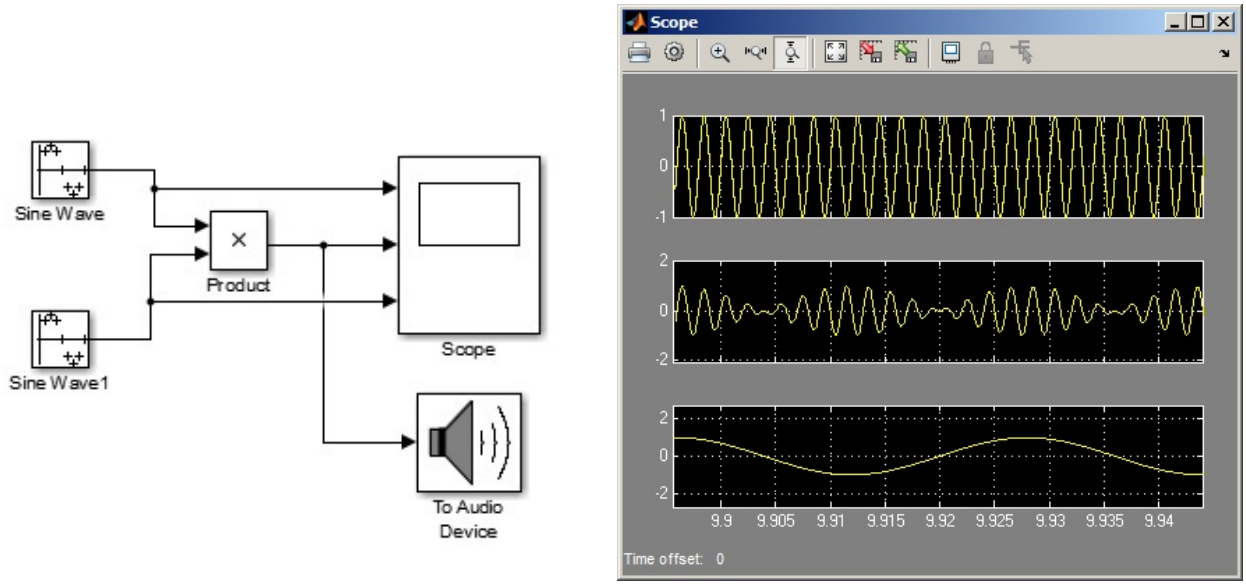
(a) On your `Simulink Scope`, Show the TA the result of adding a 1Vp, 1KHz sinusoid and a $\frac{1}{2}$Vp, 2KHz sinsoid. Use the slider gain blocks to assign the magnitudes.

2.2.4 Multiplying Two Sinusoids

Finally, another interesting operation is the multiplication of two signals. To make things resemble your old high-school days, let us multiply two sine waves. As you know, when you multiply two sine waves of different values, you end up with two sine waves again, but with a different amplitude and different frequencies (recall the $\sin(a) * \sin(b)$ formula). The resulting sine waves have frequencies that are the sum and the difference of the original frequencies. In Electrical Engineering, a multiplication by a sine (or cosine) is referred to as *modulation*. Examples of modulation are Amplitude Modulation (AM), Frequency Modulation (FM), etc.

Now build a model to try this out, as shown in figure 10.

The frequencies you will use are 31.25Hz and 500Hz. The amplitude is 1 for both. Now it would be nice to see all signals on the scope while listening to them. The way to make the multiple-input scope shown on Figure 10(b) is to click on the scope



(a) Model (b) Resulting Waveform

Figure 10. Modulation

parameters (the gear-like symbol at the top left of the scope menu), and select the number of inputs you want in the `Number of axes` field. The case shown is for 3 inputs.

Exercise 2.2.3:

(a) Put the model together, make it run. Show it to the TA and let him hear the sound.

You may want to try different frequency values. What you hear is actually an effect used in audio called a *Ring Modulator*.

3. Conclusion

In this experiment you became familiar with some extra capabilities of `Matlab` and were introduced to `Simulink`. You have tried a basic set of blocks found in `Simulink` and made your models interact with `Matlab` and with the outside world. Perhaps you can now make your own musical instrument with `Simulink`. Do this as a side project, though, because there is plenty to come in ECE216!