University of Pennsylvania
Department of Electrical and Systems Engineering
Digital Audio Basics

ESE250 Spring 2012 Lab 2: Sampling Friday, January 20, 2012

For Lab Session: Tuesday, January 24, 2012 in Detkin Lab.
Due: Monday, January 30, 2012 by noon.

Collaboration: Work in lab in teams of 2 (Different teams than lab 1). Perform individual write-ups. See course collaboration policy in the Administrative Handout.

Objective: Digitally synthesize a tone into a headphone

- Produce a digital representation of a sound wave
- Observe (with your headphones/ears and with an oscilloscope) the produced wave

Prelab Requirements: Download and read this entire lab assignment especially the Motivation section and the Theory subsection of the Lab Report Guidelines section. Remember to bring your headphones to lab with you on Tuesday.

Deliverable:

- Names of all lab group members
- Answers to all lab questions (including prelab question)
- Code or spreadsheet used to generate digital sound wave
- A brief README describing how to use your code.
- For each of the following, the file containing the digital representation of a
  - 1 second, 220Hz, Amplitude 1 wave unquantized
  - 1 second, 440Hz, Amplitude 1 wave unquantized
  - 1 second, 3,520Hz, Amplitude 1 wave unquantized

Handin: All labs will be turned in electronically through the Penn Blackboard website. Go to the assignment submission link and follow the instructions. Please submit all your files as **one zip file**. Also, remember that your writeup should be a **PDF file**.

Exit Ticket: Show your TA the generated waves on LabVIEW and on the oscilloscope.

Motivation

You learned in class that sound waves can be discretely represented as a series of numbers defining the amplitude of the wave over time. In this lab, you will write a program that generates these numbers for a pure tone (sine wave) given an amplitude, frequency, and sample rate, and then generate a LabVIEW VI to quantize and measure the signal. Finally, you will play and display the generated wave.

Lab Procedure

Digital Representation (Sampled Waveform Generator)

In this part of the lab you will develop and implement a program that produces a digital representation of a sound wave. For simplicity, we recommend you do this in a spreadsheet program like Microsoft Excel, but you are free to use any programing language. Your program will implement your $s(t)$ function from your pre-lab which should use the standard $sin()$ function.

You should implement your program so that it can take as inputs a frequency and an amplitude and generate enough points to create a 1 second wave. We will be using a sample rate of 48,000Hz so that can be hardcoded in your program.

Your program should generate as output a text file that has one point per line. A point is a decimal number between -*amplitude* and *amplitude* inclusive. Also, in order for it to be compatible with LabVIEW, make sure the extension for the output file is `.lvm`. For Excel users, you can manually create the text file by copying and pasting contents of the cells to a text file (created using your favorite text editor like Notepad or Notepad++), and then saving the file with the `.lvm` extension.

Note for Excel users

- Dragging across thousands cells is slow and will take a minute or so. It is worthwhile to plan things out so that you only have to do this once. That means, you probably want to test out your routines on a small number of samples, and only generate the large number of samples required once you know your 'program' works as intended.
- Keep in mid that we will be generating multiple samples of different frequencies.
  - One way to do this is to have a single (freq, amplitude) sample per spreadsheet and make sure that all the sample cells are all driven off a single set of configuration cells so you only have to change the few configuration cells to change all the samples.

    - Another is to have multiple columns (*i.e.* one column for each sample). If you do have multiple columns, you still want to make sure this is all setup so you can do a single drag to generate all the rows for all the columns at once.

- It will be useful to understand the difference between absolute and relevative references in Excel. (*i.e.* What's the difference between A1, $A1, A$1, and A1 ?) For more details, inside Excel, invoke Help and search for "absolute cell". The help page titled "Switch between relative, absolute, and mixed references" may be most useful.

Samples for Analysis

Generate one `.lvm` for each of the following frequencies, all at amplitude 1: 110Hz, 220Hz, 440Hz, 880Hz, 1,760Hz, 3,520Hz, 7,040Hz, 14,080Hz.

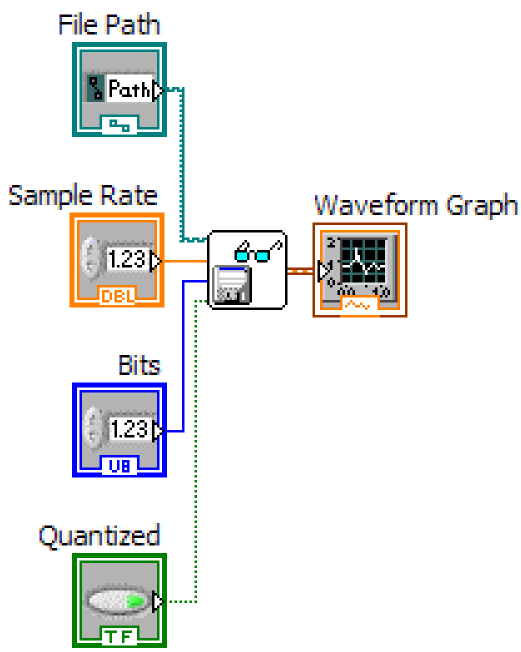
FOR QUESTIONS: Plot the points in your `.lvm` files for 220Hz, 3,520Hz, and 14,080Hz.

LabVIEW Display

Now that you have generated `.lvm` files, we will use LabVIEW to quantize and look at the generated wave. Make sure to download `readLVMFile.vi` from the lab website.

- Open LabVIEW and create a new VI
- Open the Block Diagram (`Ctrl+E` or **Window** → **Show Block Diagram**)
- In the Block Diagram, choose **Select a VI...** from the **Functions** window. Navigate until you find `readLVMFile.vi` and select it.
- To see a description of `readLVMFile.vi`'s terminals open the Context Help (`Ctrl+H` or **Help** → **Show Context Help**). It shows that `readLVMFile.vi` has two input terminals, File Path and Sample Rate and one output terminal, Waveform.
- Right click on the File Path terminal and select **Create** → **Control**.
- Create controls for the Bits and Quantized terminals as well. The Quantized switch created will let us choose whether to quantize the wave or not. The level of quantization is defined by 2^{Bits} when the quantized switch is on. Bits can range from 1 to 16.
- Right click on the Sample Rate terminal and select **Create** → **Constant**. Make sure the constant is set to 48000 to match our sample rate.
- Switch back to the Front Panel (`Ctrl+E` or **Window** → **Show Front Panel**)

- Under the Express tab of the Controls window, select the Graph Indicators, choose a Waveform Graph and place it anywhere on the Front Panel.
- Return to the Block Diagram and connect the output terminal of the `readLVMFile.vi` to the Waveform Graph. Your VI should look similar to this

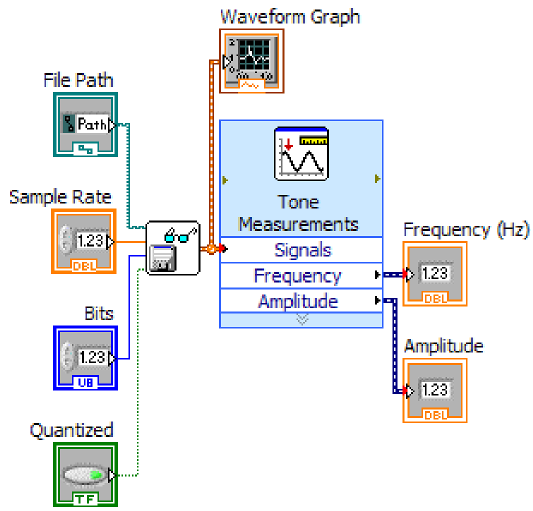


- Now you should be able to display your generated waveforms. In the Front Panel browse for one of the files you created by clicking on the yellow envelope of the File Path control (if you are having trouble clicking it, make sure no controls are selected by a dotted outline). Leave the Quantized switch off and the Bits at 16. Run the VI (`Ctrl+R` or **Operate** → **Run** or click on the Run arrow in the toolbar).

If everything worked correctly you should have seen a wave displayed in the Front Panel; however, we need to measure the wave to make sure it actually has the correct frequency and amplitude. To do this we will use one of LabVIEW's Signal Analysis tools.

- In the Block Diagram, under the Express tab of the Functions window, select Signal Analysis and place the Tone Measurements subVI in your VI.
- When the Tone Measurements configuration windows comes up, make sure Amplitude and Frequency are checked and choose OK.
- Wire the Signal terminal of the Tone Measurements to the output of the `readLVMFile.vi`, it should automatically create a branching point for the connection.

- Right click on the Amplitude terminal of the Tone Measurements and select **Create → Numeric Indicator**. Repeat with the Frequency terminal. Your VI should look something like this:



- Leaving the Quantized switch off, run the VI again for each of the files you generated.

- Confirm that the frequency and amplitude match closely what you expect for each file. If you find they don't match your expectations you should check your spreadsheet/program from the first part of this lab and make sure it's correct.

Output to Headphone

Equipment for this section

- BNC to RCA connector
- RCA to stereo cable
- 3.5mm jack splitter
- Headphones

Finally we will look at the wave on the oscilloscope and listen to it through headphones.

- In the Block Diagram, under the Express tab of the Functions window, select Output and place the Play Waveform subVI in your VI.

- In the configuration window make sure Speakers is selected.

- Plug in your headphones to the headphone output on the front of the computer and press Test Device. If sound plays, unplug your headphones and proceed with the directions. If not, follow the extra steps below:

  - Right click on the speaker icon on the right of the taskbar and select playback devices.
  - Switch which set of speakers is listed as the default speakers.
  - Delete the Play Waveforem VI and replace it with a new one.
  - Select one of the speakers (there should be two) as the Device and press Test Device again.
  - If no sound plays, select the other speaker as the Device and press Test Device
  - If sound plays unplug your headphones and proceed. If not, ask a TA for help.

- Connect the Data terminal of the Play Waveform subVI to the Waveform terminal of the `readLVMFile.vi`, again a branch should form.
- Connect the 3.5mm splitter to the headphones output on the front of the computer.
- Connect your headphones to the splitter.
- Connect the RCA to stereo cable to the splitter.
- Connect the RCA to BNC connector to input one on the oscilloscope
- Connect the red end of the RCA cable to the oscilloscope
- Turn on the oscilloscope and make sure it's set up to display (See lab 1 for more details)
- In LabVIEW, load one of your `.lvm` files and run the VI. You should have the run loop by clicking on the *Run Continuous* button on the task bar (to the right of the run button).
- With your VI running, you can turn on the Quantized switch and change the Bits to see and hear the quantized wave. Since the waves are 1 second long, you should wait at least a second for the Quantized switch and the changing Bits to have an effect on the produced wave.
- Show your TA the 1 second, 220Hz, Amplitude 1, 2^3 quantization level wave on the oscilloscope and LabVIEW (required for **Exit Ticket**)

FOR QUESTIONS: Run your VI with all other `.lvm` files you generated and examine their outputs on the headphones and oscilloscope. Make sure to look at the unquantized and quantized waves (at various quantization levels). Be sure to make note of differences that you notice in the sound and in the visual waveform.

Lab Writeup Guide

Theory

Prelab Questions:

- When digitizing a signal, why must it be quantized in space and time?
- In this lab we want to generate a sequence of values representing a **sampled** sound wave. Assume the function $s(t)$ gives you the unquantized value of the wave at sequence time t. Using the standard $sin()$ function, what is $s(t)$ for a pure tone (sine wave) of amplitude, A, and frequency, f, sampled at rate, r?
- In this lab, we will be generating 1 second samples. Given that we will sample at 48,000Hz, how many points will a 1 second sample require? How does this number change as the sampling frequency changes?

Analysis

Question 1: Looking at the **plotted version** (not LabVIEW output) of your waves, what do you notice about the shapes of the waveforms as the frequency changes? What is a good explanation for your observation?

Question 2: Describe how the waveforms produced by LabVIEW sound on the headphones and looks on the oscilloscope.

Question 3: How, if at all, are these waveforms different from the waveforms produced by the signal generator in the previous lab? (no amplitude quantization? with amplitude quantization?)

Question 4: How does the wave change (look and sound) as the amplitude quantization level changes?

Question 5: At what amplitude quantization level can you no longer hear the difference between the digital wave and the waves generated by the function generator in lab 1? At what amplitude quantization level can you no longer see the difference? Are these values the same for all frequencies?

Conclusion

Question 6: Considering the full precision waves (no amplitude quantization) produced by LabVIEW, what evidence, if any, can you observe in the output that the input provided was discretely quantized in time? Why do you think this is the case? What does that tell you about sampling and reproducing sampled waves? (*We need a very simple answer here. Think about relationship between the input—the points in your file—and the output and the fact that they are passing through a VI. You will learn more about this in later labs*)