Mini Mixer

Core Concept Instructor Set

Mini Mixer Core Concept

Audio Output Left

Audio Output Right

Channel Audio Files

Channel 1 | Volume Fader
Channel 2 | Volume Fader
Channel 3 | Volume Fader

STOP

Mini Mixer Example VI
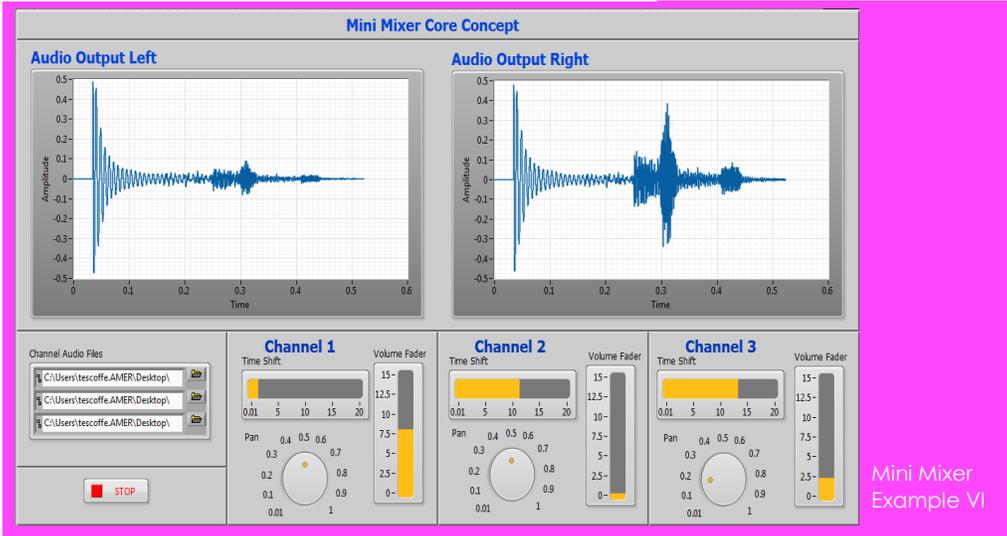
Materials:

myDAQ

Stereo Speaker/Headphones

"Speakers take many forms from those found in laptops to sound bars but the fundamental functionality stays the same."

Learn It!

The typical speaker design takes advantage of the principals of electromagnetism. As current runs through a wire, a magnetic field is produced around the wire. A coil of wire with current passing through it is called an electromagnet. The speaker design suspends a free moving electromagnet in a permanent magnet, a large cone shaped material is attached to the electromagnet in order to move air particles as the electromagnet fluctuates. The fluctuation of the electromagnet is caused by varying the current signal. If an audio signal is used to vary the current through an electromagnet then the cone attached will move air according to the amplitude and frequency of that signal. That moving air translates to sound! Speakers take many forms from inside laptops to sound bars but the fundamental functionality stays the same.

In this module, you will manipulate and mix together audio signals that will be used to drive a stereo speaker or headphones.

Build It!

Audio signals are handled the same as arrays in LabVIEW. Let's explore how to create and combine analog audio signals in LabVIEW.

Note: This module will be done entirely in LabVIEW. Though this is a step by step process, take some time to familiarize yourself with functions outside those stated in the module.

Creating an Audio Signal

Step 1: On the block diagram, place a **Basic Function Generator.vi** by right clicking in an empty area on the block diagram and navigate to *Signal Processing >> Waveform Generation >> Basic Function Generator.vi.*

Step 2: Click on that function to select it then copy and paste a second one by pressing Ctrl+C and Ctrl+V. These two functions will create the two signals you will be using for this portion of the module.

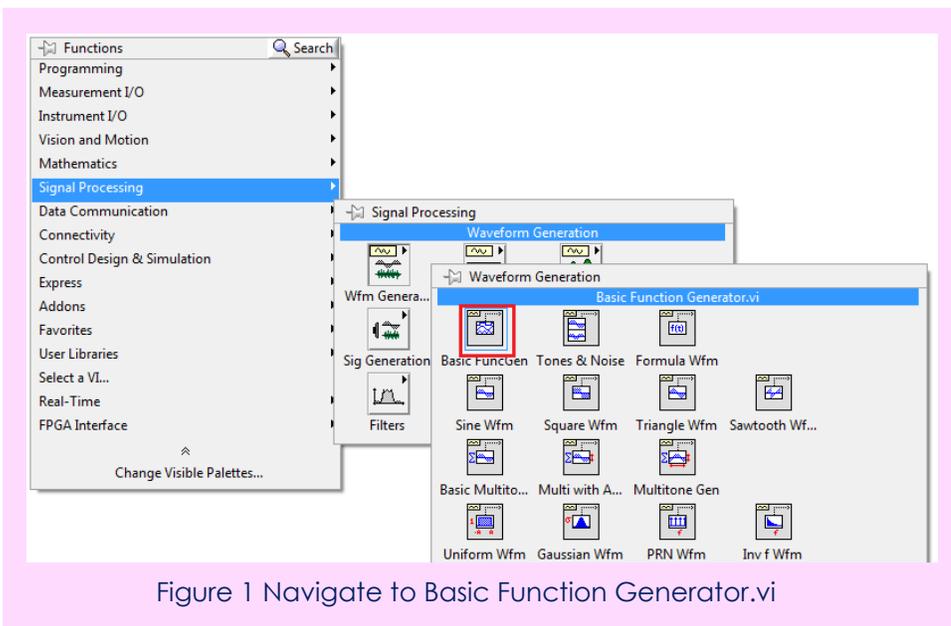Picture that shows how Task 1 should look.

Figure 1 Navigate to Basic Function Generator.vi

Step 3: Create controls for the "frequency", "phase" and "signal type" terminals on each Basic Function Generator.vi by hovering your mouse over the corresponding terminal on the function until it turns into the wiring tool, then right clicking and choosing **Create>>Control**. The Amplitude of each function can be controlled separately but it will be easier if there is one Amplitude control to interact with so for this portion of the module make one control for the amplitude and wire it to both functions.

Step 4: Create a constant for the "sampling info" terminal by following the same procedure as creating a control but instead choosing the constant option. The sample info constant contains two values, the top value is the sample rate in samples per second and the bottom value is number of samples. Change the sample rate value to '41000' and the number of samples to '4100' then wire the constant to the other Basic Function Generator.vi. Your block diagram should now resemble Figure XX.

Figure 2 Generating two signals

Guiding Questions:
- What is the significance of the sample rate being 41000 Sa/s?
- What is the maximum output amplitude that your myDAQ device can handle?
- Can you pick up the full audio range of frequencies with a single speaker? Explain why or why not and propose a solution if not.
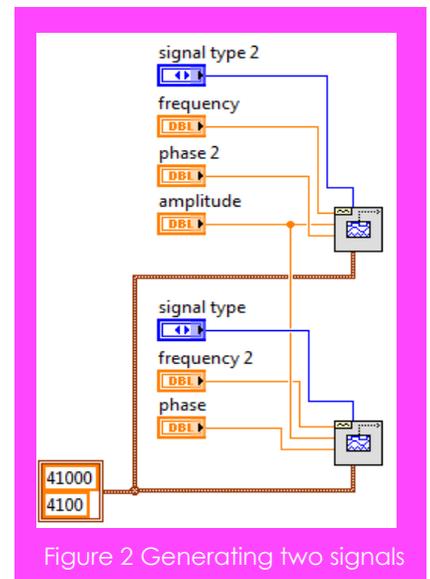
Step 5: Place an Add function on your block diagram by right clicking and navigating to **Programming>>Numeric>>Add**. Wire the signal out of both of the Basic Function Generator.vi blocks to the Add function. Adding the two outputs together will allow for you to hear both signals you've created once you output them to your myDAQ.

Step6: Place a Merge Signals function on your block diagram by right clicking and navigating to **Express>>Signal Manipulation>>Merge Signals.** The Merge Signals function combines multiple signals and represents them with one dynamic data type, we need this in order to wire into our Analog output function once we place it. Wire the output of your Add function to both of the inputs of the Merge Signals Function. This will produce a single signal that contains information for the left and right audio output for your sound.

Step 7: Place a DAQ Assistant on your block diagram by right clicking and navigating to **Measurement I/O>>NI-DAQmx>>DAQ Assistant.** This should open a configuration window once it is placed down. In that window select **Generate Signals>>Analog Output>>Voltage**. This should lead you to another configuration window which should list your myDAQ device if it is connected and has been detected by your computer, as well the channels it has that can generate an analog output. Click on audioOutputLeft, press and hold Shift then click audioOutputRight to select the two channels, now click Finish.

Step 8: You should now be at a configuration window that looks similar to Figure XX. Change the Max and Min values for both VoltageOut_0 and VoltageOut_1 to **2 and -2** respectively. Change the generation mode to **Continuous Samples** and click Ok. You will be prompted to automatically place a while loop around your DAQ Assistant function, choose No. Wire the output of the Merge Signals function to the data input of the DAQ Assistant.

Step 9: Right click on your block diagram and navigate to **Programming>>Structures>>While Loop**. This should change how your cursor looks to indicate you can now select the area that will be enclosed in your while loop. Click and drag around the code you have created up to this point, this should place a while loop around it. Hover your mouse over the

Loop condition terminal 🔴 , right click and select Create Control to create the stop button for the VI.
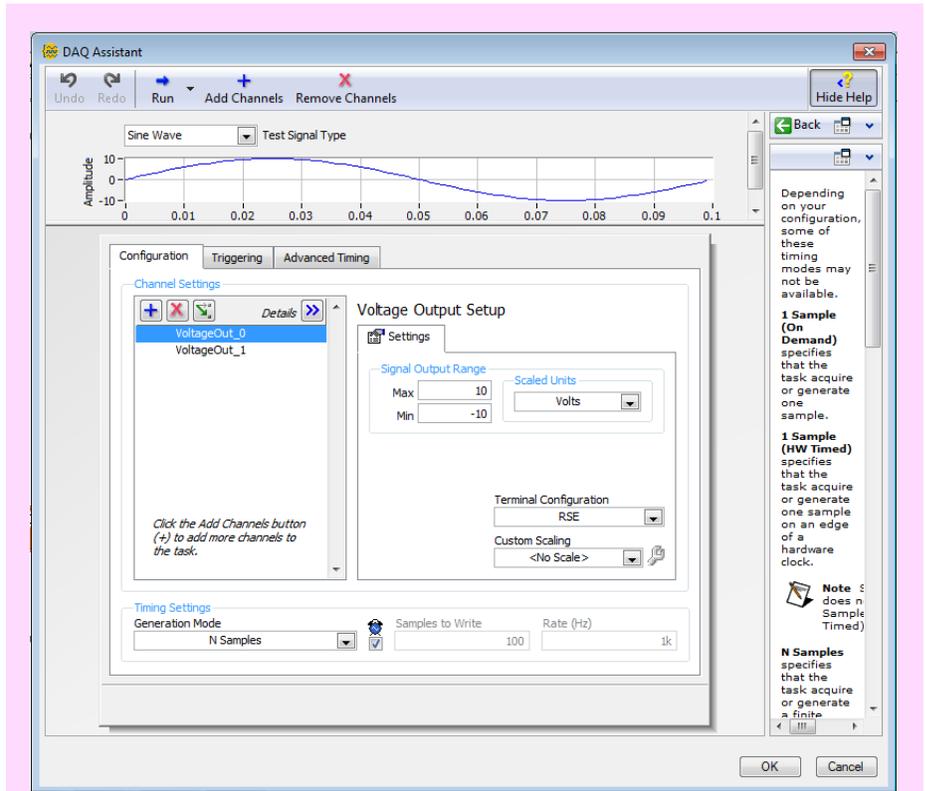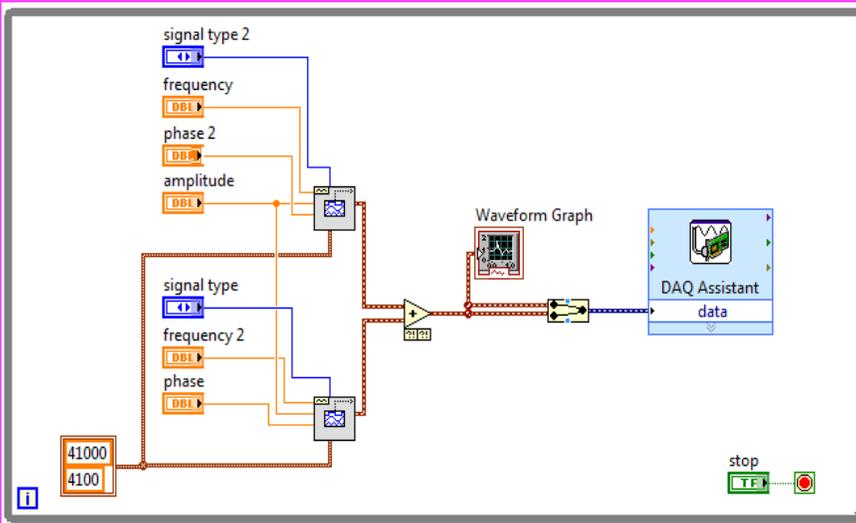

Figure 4 DAQ Assistant task configurations

Figure 3 Completed block diagram adding signals

Step 10: Switch to the front panel by pressing Ctrl + E, and place a Waveform Graph indicator by right clicking and navigating to **Silver>>Graph>>Waveform Graph.** Switch back to the block diagram and wire the output of the Add function to the Waveform Graph indicator. Your block diagram should now resemble Figure XX. Switch back to the front panel, right click on the Waveform Graph indicator and select **X Scale>> AutoScale X** in order to disable auto scaling. Double click on the right most number on the X axis to highlight it and change it to **0.005**.

Plug your speaker or headphones into the Audio Out of the myDAQ, set a value of 0.0 on the amplitude control on your front panel and run your VI. Slowly increment the amplitude control until the sound level on your speakers or headphones is comfortable. Examine the output signal and take note of the sound it is making. Vary all of the controls on your front panel and see how the changes affect both the signal being seen and heard.

Guiding Questions:
- What is the maximum amplitude you can achieve for each individual signal if you are going to be adding them together and outputting on the AudioOut channel of the myDAQ? What happens as you increase the amount of signals added?
- Explain how to create the effect of a sound coming from a specific direction. What more does that involve?

What's happening?

If two signals have the same frequency and phase then the sum of the two will result in a signal with the same frequency and phase but amplitude that is the sum of the two signal's amplitudes. As the two frequencies are changed the higher frequency signal starts to become superimposed on the amplitude of the lower. What is heard is still two distinct sounds coming from the speaker or headphones. As the phase of the two signals start to change the signals begin to interfere with one another. Two signals with the same amplitude and frequency that are 180 degrees out of phase, for example, will cancel one another out. The signal data is represented by a waveform which defines an array of values that represents the amplitude of the signal at a specific time instance, a period between each amplitude point (dt) and the initial time of the waveform (t_0). The data can be interpreted as array element zero corresponding to t_0 and all subsequent array elements corresponding to ($t_0 + Ndt$) where N is the index of the array element.

Explore the Mini Mixer

Step 1: Open the Mini Mixer_Courseware.vi and load three audio signals into the Channel Audio Files control. Example files can be found in the Example Audio folder of the Mini Mixer main folder or you may find and load audio files of your own. Choose your myDAQ device from the Device Name control. Verify that all of the Volume Fader controls are at 0.0 and run the VI.

Step 2: Vary the controls and observe how the Left and Right Audio signals change as well as how the sound changes. Navigate to the block diagram, open the Analog Output Manipulation.vi and examine its functionality.
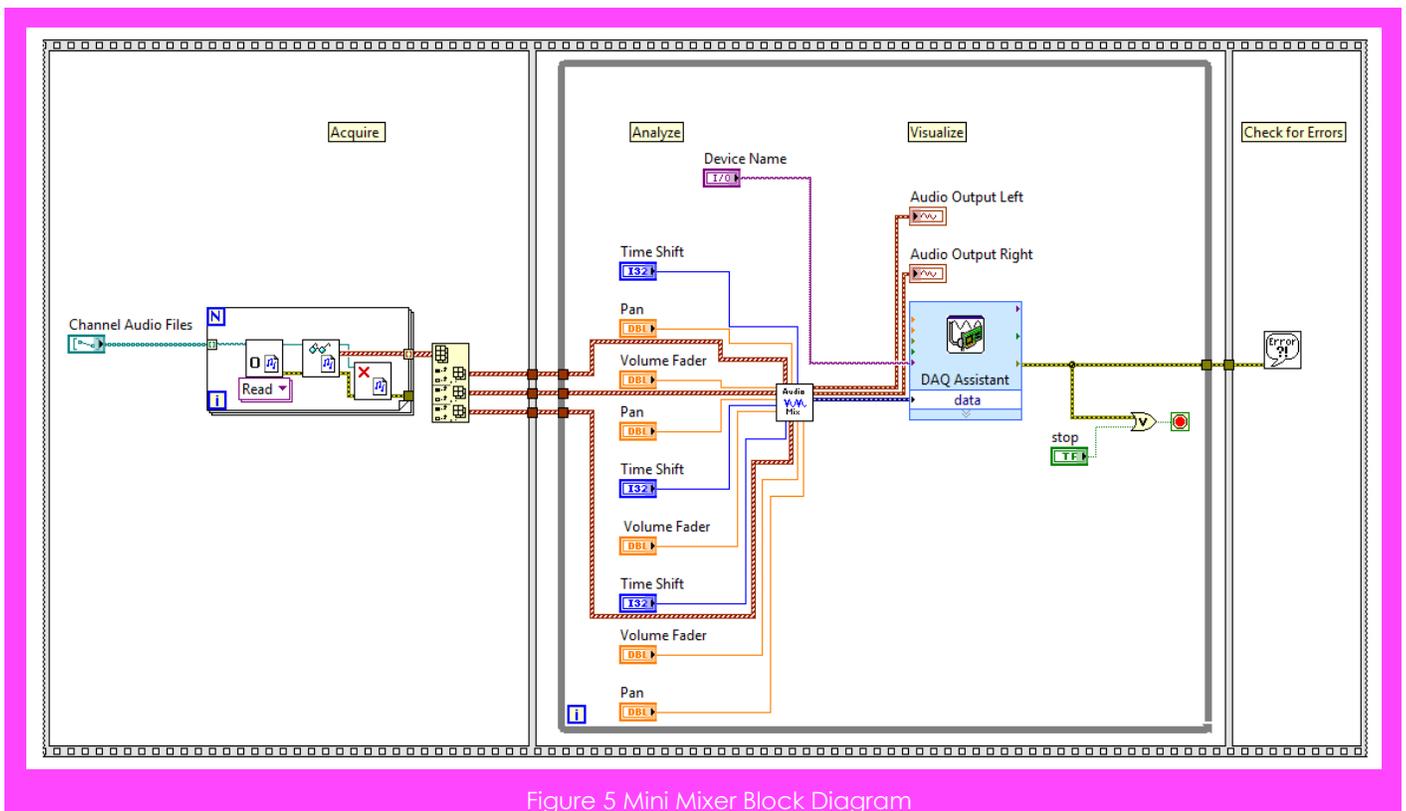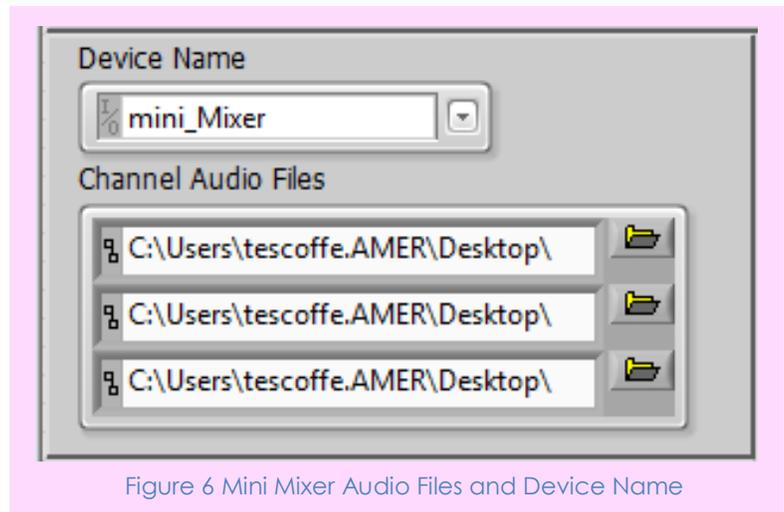
Figure 6 Mini Mixer Audio Files and Device Name

Figure 5 Mini Mixer Block Diagram

Expand it!
- Use two potentiometers and the analog inputs of the myDAQ to implement the volume and pan controls of the Mini Mixer in hardware.
- Monitor the volume levels of each channel and use colored LEDs to indicate when a channel passes a volume threshold (i.e when a the volume of a channel passes 0.8 on the volume slider)

Textbook References

Include a list of sections of textbooks this project touches on. Include the title, chapter and section name as well as page numbers. This section can also contain a listing of sections that can be taught in parallel with or after this project.