

Acoustic Signal Acquisition and Frequency Analysis

Case Study

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1. Overview

Short description: This **Case study** explores how a microphone module with an integrated amplifier can be used to capture acoustic signals and convert sound pressure variations into measurable electrical voltage signals. Participants generate tones at different frequencies using a smartphone, observe the captured signals in the time domain, and analyze their frequency components to investigate the frequency response of the microphone system. The **Analog Discovery Product Line** (AD 3 or AD Studio / Studio MAX, hereby **Device**) supplies power to the microphone module and simultaneously acquires the output signal for visualization and analysis using the oscilloscope and spectrum analyzer.

Objectives

- Acquire and observe microphone output signals in the time domain using the **Device** oscilloscope.
- Analyze the frequency components of captured acoustic signals using the **Device** spectrum analyzer.
- Investigate how changing the generated sound frequency affects the measured signal and frequency response of the microphone system.

Software & Hardware Requirements

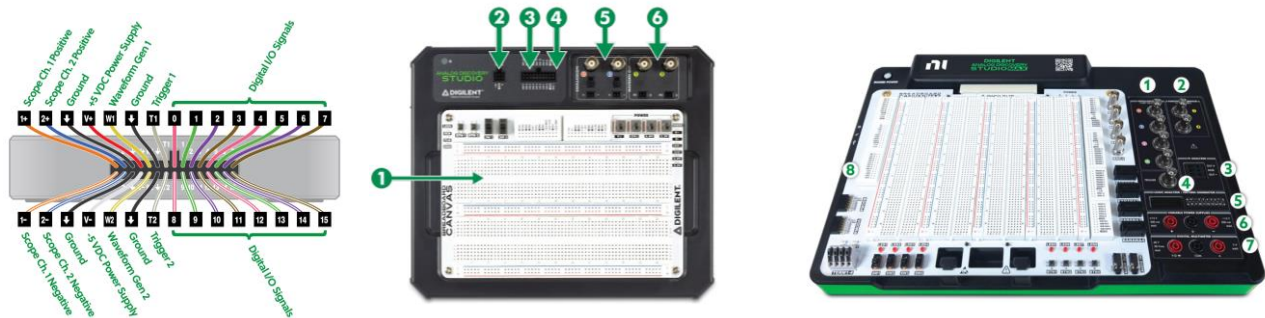
Software		Version
Digilent WaveForms (Runtime + SDK)		Latest (64-bit)
NI LabVIEW		2025 64-bit
WaveForms Toolkit for LabVIEW		Latest
Hardware		Quantity
Digilent Analog Discovery Product Line		
Option 1	Digilent Analog Discovery 3	1
Option 2	Digilent Analog Discovery Studio	1
Option 3	Digilent Analog Discovery Studio MAX	1
List of Components		Quantity
Canvas Breadboard		1
Microphone Module with Integrated Amplifier		1
Tone generator device ¹		1
Jumper wires		-
BNC to BNC cable ²		1
Banana to Banana cable		1

¹ A smartphone with a tone generator application can be used as the tone generator device.

² BNC cable required only for the Analog Discovery Studio MAX.

2. Hardware Setup

This section describes the **Digilent Analog Discovery Product Line** hardware setup for **Case studies**. Before assembling the measurement circuit, it is essential to understand the pinout and available connections of the **Devices**. **Figure 1** highlights the pinouts and instrumentations of the Digilent Analog Discovery product line.³



a. Analog Discovery 3

- Power Supplies
- Logic Analyzer and Pattern Generator
- Triggers
- Oscilloscope
- Waveform Generator

b. Analog Discovery Studio

1. Breadboard Canvas
2. Power Supplies
3. Logic Analyzer and Pattern Generator
4. Triggers
5. Oscilloscope
6. Waveform Generator

c. Analog Discovery Studio MAX

1. Oscilloscope
2. Waveform Generator
3. IV Analyzer
4. Trigger
5. Logic Analyzer and Pattern Generator
6. Variable Power Supplies
7. Digital Multimeter
8. Breadboard Canvas

Figure 1. Digilent Analog Discovery Products Pinout and Instrumentations

Table 1 provides a common reference for physical channels' names across the **Digilent Analog Discovery product line**. The **Case studies** are designed to work with all supported devices, so the generic channels' names shown in the first row are used consistently in all circuit diagrams and wiring instructions.

Table 2 provides a common reference for device names and instrument identifiers used in NI LabVIEW with the WaveForms Toolkit across the **Digilent Analog Discovery product line**. The Case studies are designed to work with all supported devices; therefore, the generic instruments and

³ For detailed electrical specifications, pin descriptions, and operating limits, always refer to the official datasheets of each device (Analog Discovery 3, Analog Discovery Studio, and Analog Discovery Studio MAX). The latest datasheets are available on the Digilent website:

- [Analog Discovery 3](#)
- [Analog Discovery Studio](#)
- [Analog Discovery Studio MAX](#)

channels identifiers shown in Table 2 are used consistently throughout the LabVIEW block diagrams and configuration steps.

Table 1. Physical Channels Mapping for Digilent Analog Discovery Devices.

<i>Channel Device</i>	<i>Scope Ch.1 Positive</i>	<i>Scope Ch.1 Negative</i>	<i>+Vs</i>
AD 3	Scope Ch.1 Positive (Pin 1+)	Scope Ch.1 Negative (Pin 1-)	+VDC Power Supply (Pin V+)
AD Studio ⁴	OSCILLOSCOPE 1 (Pin + or BNC Positive Cable)	OSCILLOSCOPE 1 (Pin - or BNC Negative Cable)	+5.0V (on canvas board)
AD Studio MAX ⁴	OSCILLOSCOPE 1 (BNC Positive Cable)	OSCILLOSCOPE 1 (BNC Negative Cable)	+5.0V (on canvas board)

Table 2. Device Names and Instruments Identifiers in NI LabVIEW (WaveForms Toolkit).

<i>Identifier Device</i>	<i>Device name</i>	<i>Scope Ch.1</i>	<i>Scope Ch.2</i>	<i>PS Channel</i>
AD 3	Discovery3	mso/1	mso/2	ps/+
AD Studio ⁴	DiscoveryStudio			
AD Studio MAX ⁴	ADS_Max			

⁴ For the Analog Discovery Studio and Analog Discovery Studio MAX, ensure the devices are powered by turning ON the main power switch before use.

3. Background Information

A microphone converts acoustic pressure variations into an electrical voltage signal that changes over time. In this **Case study**, a microphone module with an integrated amplifier captures sinusoidal acoustic signals generated at different frequencies. The amplified output is acquired by the **Device** and analyzed in both the time and frequency domains. The **Oscilloscope** displays the measured signal in the time domain, showing how the signal amplitude changes with time. For a periodic sinusoidal signal, the frequency can be determined from the measured period (T):

$$f = \frac{1}{T}.$$

The **Spectrum** instrument represents the same signal in the frequency domain and shows the individual frequency components contained in the measured waveform. This transformation is performed using the **Fast Fourier Transform (FFT)**, an efficient algorithm used to calculate the frequency content of a sampled signal. The FFT converts a sequence of time-domain samples ($x[n]$) into frequency-domain values ($X[k]$):

$$X[k] = \sum_{n=0}^{N-1} x[n] e^{-j2\pi kn/N}.$$

The FFT output is divided into discrete frequency intervals called **frequency bins**. The frequency resolution of the spectrum depends on the sampling frequency (f_s) and the number of acquired samples (N):

$$\Delta f = \frac{f_s}{N}.$$

A smaller (Δf) provides better separation between closely spaced frequency components. According to the **Nyquist sampling criterion**, the sampling frequency must also be greater than twice the highest frequency contained in the measured signal:

$$f_s > 2f_{max}.$$

For a sinusoidal acoustic signal, the Spectrum typically shows a dominant peak at the generated frequency. Additional peaks may appear because of harmonics, microphone or amplifier distortion, environmental noise, or characteristics of the acoustic source. The overall background level visible in the spectrum represents the **noise floor** of the measurement. When a finite portion of a signal is analyzed using the FFT, discontinuities at the beginning and end of the acquired data can cause **spectral leakage**, where signal energy spreads into neighboring frequency bins. Window functions can be applied before the FFT to reduce this effect and improve the interpretation of the spectrum.

By generating tones at different frequencies while maintaining approximately the same sound-source position and volume, participants can compare the measured signal amplitude at each frequency. This allows the frequency-dependent behavior of the complete acoustic measurement system-tone generator, speaker, air path, microphone, and integrated amplifier-to be investigated.

4. Circuit Design – step by step instructions

4.1 Description

The measurement circuit is based on a microphone module with an integrated amplifier used to convert acoustic signals into measurable electrical voltage variations. The microphone module is powered by the **Device +Vs** supply and **GND**. The amplified analog output of the microphone module, labeled **A0**, provides a voltage signal corresponding to the detected sound. The output signal is monitored by connecting **Scope Ch.1 Positive** to the microphone module **A0** terminal and **Scope Ch.1 Negative** to **GND**. This configuration allows the **Device** to acquire the amplified analog signal produced by the microphone module.

Important! Refer to **Table 1** to determine the exact physical pin, connector, or cable that corresponds to each generic connection on the specific Digilent Analog Discovery device being used. The remaining rows in the table show how each generic channel maps the actual hardware connections for each device.

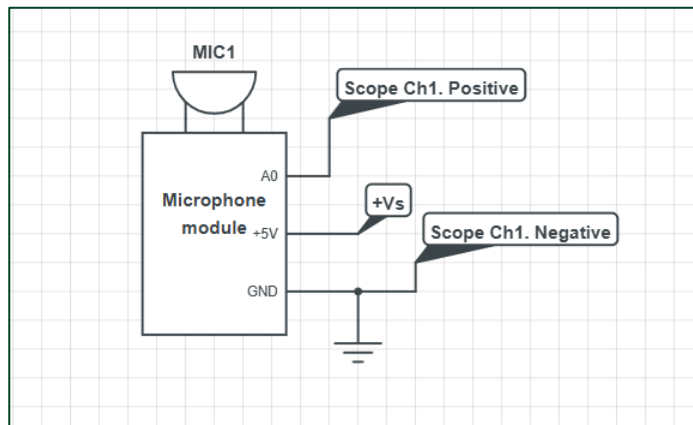


Figure 2. Microphone Module Acoustic Signal Measurement Circuit for Device

Important! For the Analog Discovery 3 (AD 3), use the programmable positive power supply output and set it to +5V to power the microphone module.

For the Analog Discovery Studio and Analog Discovery Studio MAX, use the fixed +5V supply available on the Canvas board to power the microphone module.

4.2 Step by step instructions⁵

1. Place the **microphone module** on the **Canvas breadboard**⁶.
2. Connect the **+5V** terminal of the microphone module to the **Device +Vs** supply.
3. Connect the **GND** terminal of the microphone module to the **Device GND**.
4. Connect **Scope Ch.1 Positive** to the analog output terminal **A0** of the microphone module.
5. Connect **Scope Ch.1 Negative** to **GND** to provide a common reference for the analog signal measurement.

⁵ Given on Figure 2.

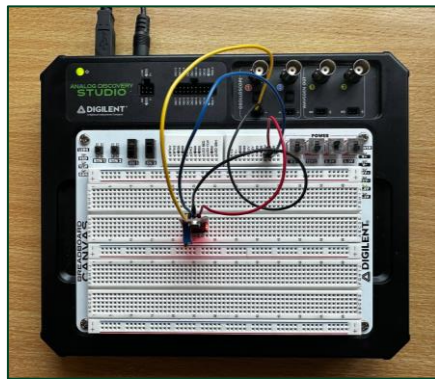
⁶ Use external breadboard in the case of Analog Discovery 3.

4.3 Assembled Circuit

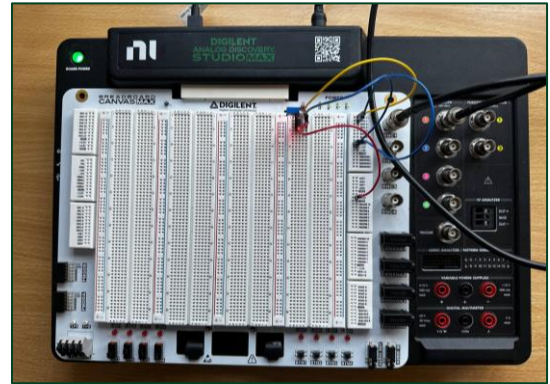
Figure 3 presents complete experimental circuits on different Digilent devices.



a. Analog Discovery 3



b. Analog Discovery Studio



c. Analog Discovery Studio MAX

Figure 3. Assembled circuits for different Devices

5. Software Setup - step by step instructions

5.1 Data Visualization and Analysis in WaveForms

Step 1. Launch WaveForms and connect the device

- 1.1. Connect the **Device** to the computer via USB.
- 1.2. Open the **Digilent WaveForms** software.
- 1.3. Verify that the device is detected and shown as *Currently Selected* in the **Device Manager** (Settings -> Device Manager).

Step 2. Configure the Power Supply in WaveForms⁷

- 2.1. Open the **Supplies** instrument in **WaveForms**.
- 2.2. Enable the **Positive Supply (V+)**.
- 2.3. Set the output voltage to **+5.000 V DC**.
- 2.4. Enable the **Master Enable** switch to activate the power output.
- 2.5. Keep the **Negative Supply (V-)** disabled.

Figure 4 shows the configuration of the **Device** power supply using the Digilent WaveForms environment.

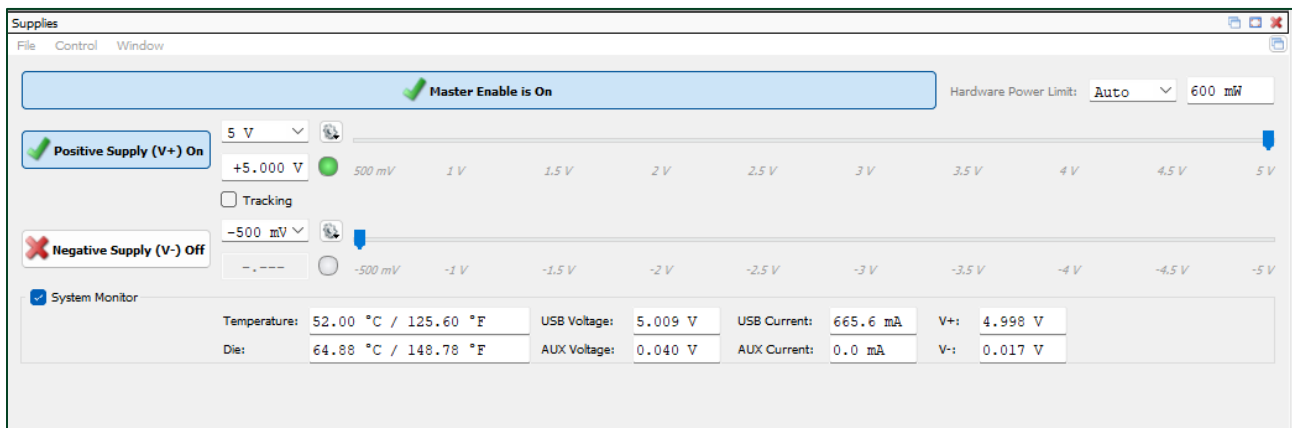


Figure 4. Power Supply Configuration in WaveForms Software

Step 3. Generate and acquire an acoustic signal using the Oscilloscope⁸

- 3.1. Position the **Tone Generator Device** approximately 1-5 cm from the microphone module.
- 3.2. Configure the **Tone Generator Device** to generate a sinusoidal acoustic signal at 1 kHz.
- 3.3. Set the output volume of the **Tone Generator Device** to a high level and keep this volume unchanged during the measurements.
- 3.4. Open the **Scope** instrument in **WaveForms**.
- 3.5. Enable Channel 1.
- 3.6. Adjust the **Range** so that the microphone output signal fits clearly on the screen. A starting value of approximately **1 V/div** can be used and adjusted if required.
- 3.7. Adjust the **Time Base** to approximately **500 μs/div** so that several periods of the 1 kHz signal are clearly visible.

⁷ This step is mandatory for the Digilent Analog Discovery 3. Skip if you are using another device.

⁸ In the case of AD Studio turn the 5.0V & V± switch on. In the case of AD Studio MAX turn Board Power on.

3.8. Run the **Scope** and observe the amplified analog signal produced by the microphone module.

Figure 5 shows an example of the acoustic signal captured by the microphone module and displayed using the Scope instrument in Digilent WaveForms. The periodic waveform represents the electrical signal generated by the microphone in response to the sinusoidal acoustic tone.

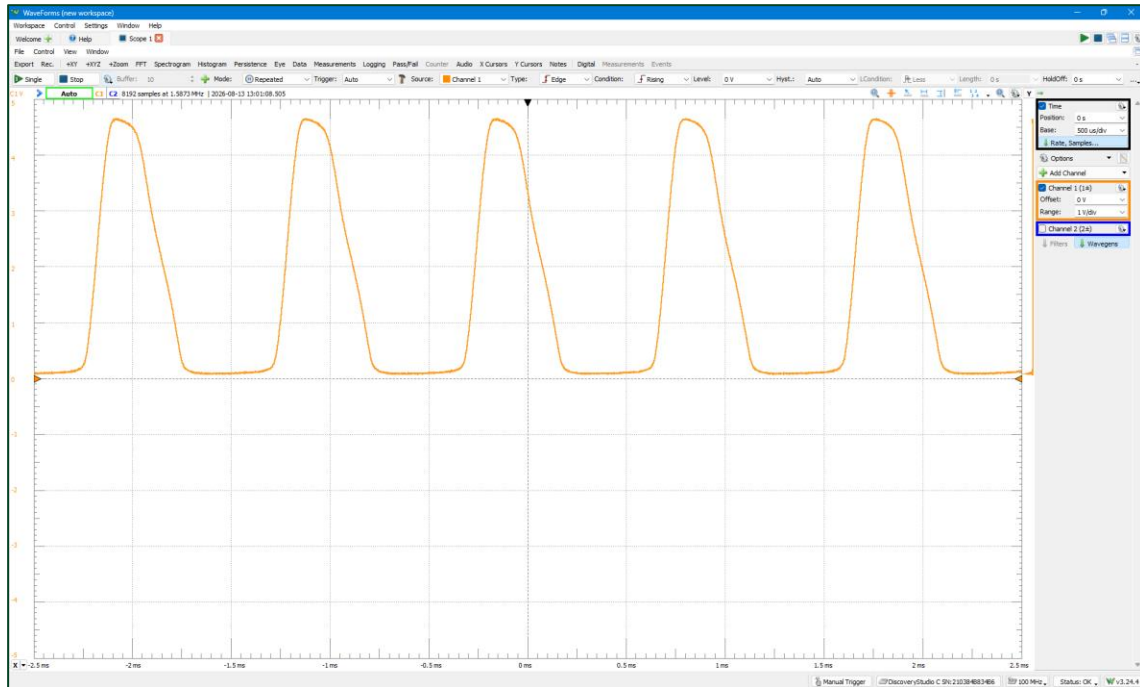


Figure 5. Microphone Output Signal Displayed in WaveForms

Step 4. Measure the signal parameters using the Oscilloscope

- 4.1. Use the cursor measurement tools in the Scope window to measure one complete period (T) of the captured waveform.
- 4.2. Calculate the signal frequency using $f = \frac{1}{T}$.
- 4.3. For a generated frequency of 1 kHz, the measured period should be approximately $T \approx 1$ ms.
- 4.4. Click **Measurements** from the top toolbar. Select **Add**, then choose **Defined Measurements**.
- 4.5. In the **Add Measurements** window, select **Channel 1** → **Horizontal** → **Frequency**, and click **Add** to display the measured signal frequency.
- 4.6. Add the **Peak-to-Peak** measurement for **Channel 1** to observe the signal amplitude.
- 4.7. Add the **RMS** measurement for **Channel 1** to determine the effective voltage level of the captured acoustic signal.
- 4.8. Compare the frequency calculated from the measured period with the frequency displayed in the **Measurements** window and with the frequency configured on the **Tone Generator Device**.

Figure 6 shows an example of the microphone output signal measured using the Scope instrument in WaveForms. The displayed waveform is captured from the microphone module while a sinusoidal acoustic tone is generated by the Tone Generator Device. The cursor tools are used to determine the signal period, while the Measurements panel displays the automatically calculated frequency, peak-to-peak voltage, and RMS voltage. These measurements allow the time-domain characteristics of the captured acoustic signal to be compared with the settings of the generated tone.

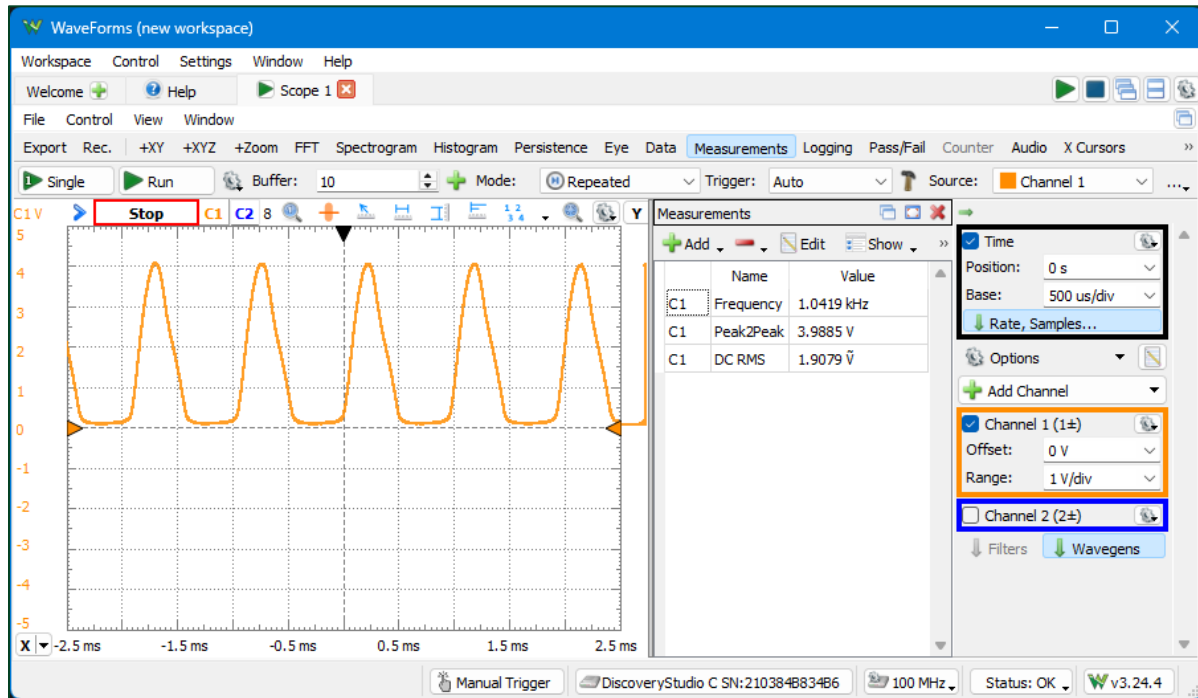


Figure 6. Microphone Output Signal Period and Voltage Measurements Using WaveForms Oscilloscope

Step 5. Analyze the captured signal using the Spectrum instrument

- 5.1. Open the **Spectrum** instrument in **WaveForms**.
- 5.2. Enable Channel 1.
- 5.3. Configure the **frequency range** so that the generated acoustic signal and its relevant frequency components are clearly visible. For this experiment, a frequency range of approximately **0 Hz to 2 kHz** can be used.
- 5.4. Start the **Spectrum** acquisition.
- 5.5. Observe the frequency spectrum of the microphone output signal.
- 5.6. Identify the dominant spectral peak. For a 1 kHz generated sinusoidal signal, the main peak should appear approximately at $f_{peak} \approx 1 \text{ kHz}$.
- 5.7. Use the Spectrum cursor or peak measurement tools to determine the frequency and amplitude of the dominant spectral component.
- 5.8. Compare the frequency obtained from the **Spectrum** with:
 - the frequency configured on the **Tone Generator Device**;
 - the frequency calculated from the period measured using the **Oscilloscope**;
 - the frequency measurement displayed by **WaveForms**.

Figure 7 shows the frequency-domain representation of the microphone signal using the Spectrum instrument. The dominant peak corresponds to the fundamental frequency of the acoustic signal generated by the Tone Generator Device.



Figure 7. Frequency Spectrum of the Microphone Output Signal

Step 6. Repeat the measurements using different generated frequencies

- 6.1. Keep the Tone Generator Device volume, distance from the microphone, and orientation unchanged.
- 6.2. Change the generated sinusoidal frequency to 2k Hz (or other value).
- 6.3. Repeat Step 4 to measure the signal parameters using the Oscilloscope, including period, frequency, peak-to-peak voltage, and RMS voltage.
- 6.4. Repeat Step 5 to analyze the same signal using the Spectrum instrument and determine the dominant spectral peak.

5.2 Data Visualization and Analysis Using NI LabVIEW WaveForms Toolkit

Step 1. Launch NI LabVIEW and connect the Device

- 1.1. Connect the **Device** to the computer via USB.
- 1.2. Open **Digilent WaveForms** and verify that **Device** is detected and shown as *Currently Selected* in the **Device Manager** (Settings -> Device Manager), then close **Digilent WaveForms** software⁹.
- 1.3. Open the **NI LabVIEW** software.
- 1.4. Create a new NI LabVIEW VI (**File->New VI**)

⁹ Before running NI LabVIEW, please ensure that the Digilent WaveForms software is closed.

Step 2. Create Block Diagram and Front Panel in NI LabVIEW

2.1. Modify the block diagram as shown in **Figure 8.1** and **Figure 8.2**.

Figure 8.1 shows the LabVIEW block diagram code applicable only to ADS and ADS Max devices.

Figure 8.2 shows the LabVIEW block diagram code applicable only to the AD 3 device.

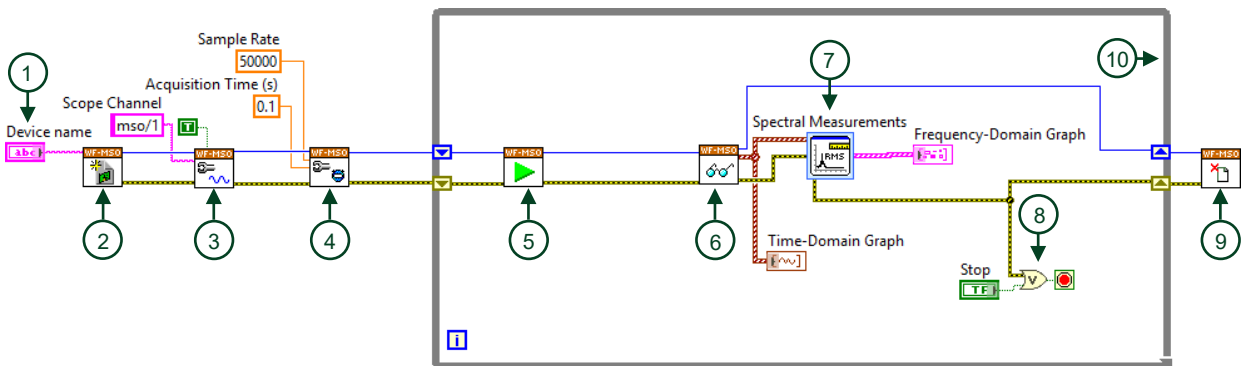


Figure 8.1. VI Block Diagram for ADS and ADS MAX

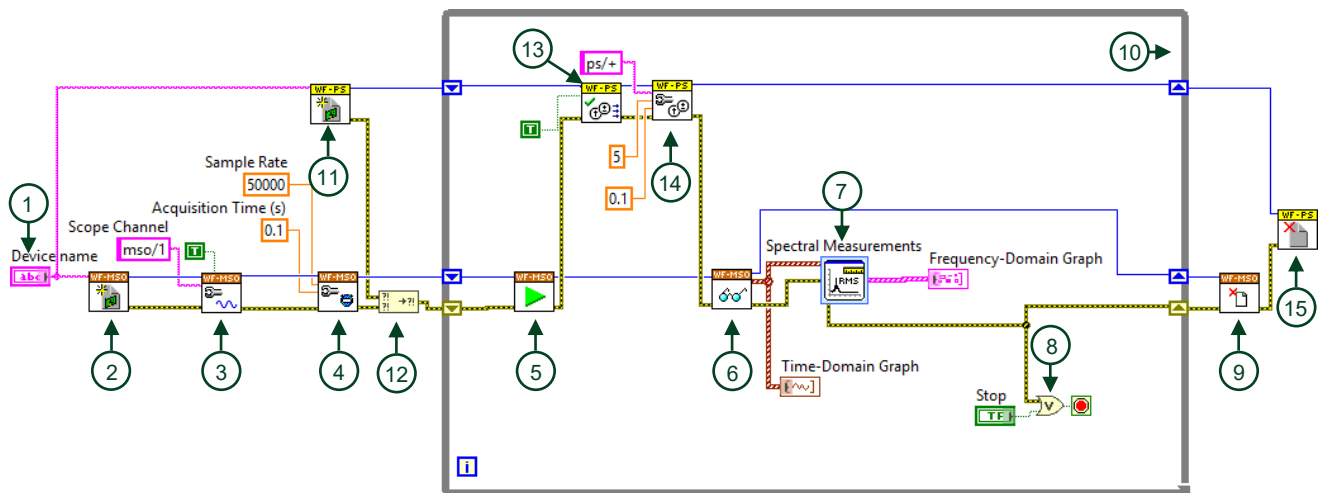


Figure 8.2. VI Block Diagram for AD 3

1. **Device Name** - specify the name of your **Device**. Use **Table 2** for your **Device**.
2. **WF-MSO Initialize** - initializes the oscilloscope instrument and establishes communication with the **Device**.
3. **WF-MSO Configure Analog Channel** - configures **Scope Ch.1** for analog signal acquisition. Specify the appropriate **Scope Channel** according to Table 2 and connect **Enable Channel** to a **True** constant.
4. **WF-MSO Configure Timing** – configures the timing parameters of the oscilloscope acquisition. Set **50,000 samples/s** and the **Acquisition Time to 0.1 s**.
5. **WF-MSO Run** – starts the oscilloscope acquisition.

6. **WF-MSO Read** – transfers the acquired microphone signal from the **Device** to NI LabVIEW. Connect the **Analog Waveforms** output to the **Time-Domain Graph** to visualize the captured acoustic signal. The same acquired waveform is also connected to the **Spectral Measurements** function for frequency-domain analysis.
 7. **Spectral Measurements** – performs FFT-based spectral analysis of the acquired microphone waveform. Configure the function to calculate the **FFT magnitude (RMS)** and connect the resulting signal to the **Frequency-Domain Graph**. The graph displays the frequency components contained in the captured acoustic signal and allows the dominant frequency generated by the Tone Generator Device to be observed.
 8. **OR** – allows the While Loop to stop either when the **Stop** button is pressed or automatically when an error occurs.
 9. **WF-MSO Close** – closes the oscilloscope session and releases the associated Device resources.
 10. **While Loop** – enables continuous acquisition and analysis of the microphone signal. During each iteration, a new waveform is acquired, displayed in the time domain, and processed to obtain its frequency spectrum.
 11. **WF-PS Initialize** – initializes the programmable power supply of the **AD 3**.
 12. **Merge Errors** – combines the error information from the oscilloscope and power-supply initialization paths into a common error path.
 13. **WF-PS Enable All Outputs** – enables the programmable power-supply output. Specify the positive power-supply channel (**ps/+**) and connect the enable input to a **True** constant.
 14. **WF-PS Configure Voltage Outputs** – configures the positive power-supply output used to power the microphone module. Set the output voltage to +5V and the current limit to 0.1 A.
 15. **WF-PS Close** – closes the programmable power-supply session and resets the power-supply resources.
-

- 2.2. Modify the front panel shown in **Figure 9**. Inside **Device name** String Control write your corresponding **Device** name from **Table 2**.
- 2.3. Save the VI.

Step 3. Run the code and perform measurements in LabVIEW

- 3.1. Run the code by clicking on the **Run button**.
- 3.2. Set the **Tone Generator Device** to generate a **1 kHz sinusoidal signal** and position it at the predefined distance from the microphone. Keep the tone generator volume and position unchanged during the measurements.
- 3.3. Observe the acquired microphone signal on the **Time-Domain Graph** and the corresponding frequency spectrum on the **Frequency-Domain Graph**.
- 3.4. Use the **Graph Cursor** tools on the **Time-Domain Graph** to measure the signal amplitude and the time difference between two consecutive waveform peaks. The time difference measured corresponds to the signal period (T).
- 3.5. Calculate the measured signal frequency using $f = \frac{1}{T}$.
- 3.6. For a generated frequency of **1 kHz**, the measured period should be approximately $T \approx 1$ ms.

- 3.7. Use the **Graph Cursor** tools on the **Frequency-Domain Graph** to locate the dominant spectral peak and determine its frequency and amplitude. Compare the measured dominant frequency with the frequency configured on the Tone Generator Device.
- 3.8. Change the generated tone frequency while keeping the **volume, distance, and orientation of the Tone Generator Device unchanged**. Repeat the time-domain and frequency-domain measurements for different frequencies. For each generated frequency, use the **Time-Domain Graph** to measure the signal amplitude and period (T), calculate the corresponding frequency, and use the **Frequency-Domain Graph** to determine the dominant FFT frequency and its amplitude.
- 3.9. Compare the measured values for the different generated frequencies and observe how increasing the acoustic frequency decreases the waveform period and shifts the dominant spectral peak along the frequency axis.

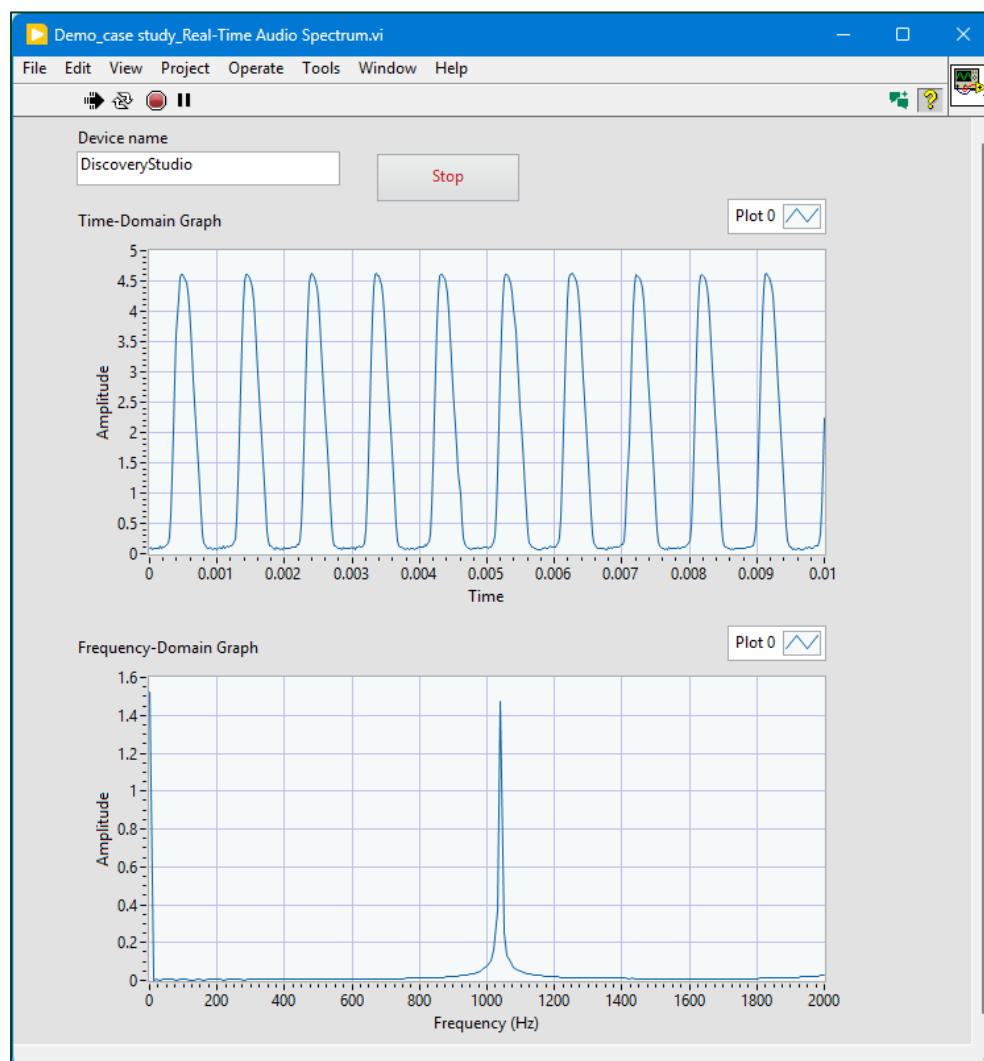


Figure 9. VI Front Panel