Testing Microphone Arrays with the APx PDM 16 Module

by Joe Begin

About This Technote

In late 2019, Audio Precision released PDM 16 – a new digital input module for its APx500 Series audio analyzers. With its unique capability to analyze audio signals from 1 to 16 PDM devices simultaneously, PDM 16 was designed to support manufacturers of digital MEMS microphones and engineers working with PDM microphone arrays. In this Technote, we discuss microphone array measurements.

About PDM 16

PDM 16 (Figure 1) is an input-only module that supports audio measurement of up to 16 PDM devices (two per data line, with eight data lines). Key specifications, primarily chosen to support digital MEMS microphone testing, are:

- Bit Clock: 128 kHz to 24.6 MHz
- Decimation Rates: 1/N, (N = 32, 64, 128, 256 or 512)
- Clock Master or Slave
- Vdd Supply (0 – 3.6 V)
- Logic Levels: 0.80 – 3.3 V

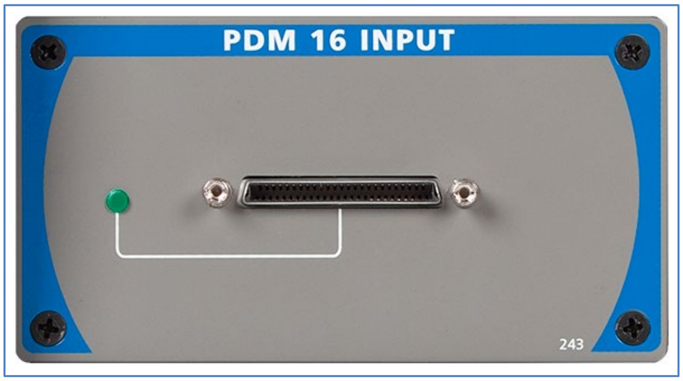

Figure 1. The front panel of the APx PDM 16 Module.

Connections to the PDM 16 module are made via a remote interface pod (Figure 2) that can be located up to 10 meters away from the analyzer (for convenience when working in an anechoic chamber). The pod has a small footprint, a low profile and is acoustically silent. Connections to the pod are via a 40-pin 2-row, 0.1-inch pitch square pin connector, typically used with a flat ribbon cable. It has LEDs to indicate when the PDM 16 module is active and when the pod is connected to the module and powered on.

Figure 2. The PDM 16 remote interface pod.

Users who need additional PDM features, such as PDM output with optional jitter generation, power supply rejection measurement or additional decimation rates should consider the APx PDM module, which provides 2-channel (one data line) PDM output and input. Note: Both the PDM module and PDM 16 module can be installed in any modular APx analyzer and used simultaneously.

Conducting the Microphone Array Tests

Microphone Array Test Fixture

The microphone array featured in this Technote is on a small printed circuit board that AP developed as a test fixture for internal testing of the PDM 16 module (Figure 3). The fixture has 16 ST Micro model MP34DT05-A PDM mics mounted in a straight line along one edge of the board with a constant spacing of 6.6 mm (0.260 in) between microphones. It also has the same connector as the PDM 16 remote pod, and connects to the pod via a 40-conductor flat ribbon cable.

With 16 microphones mounted in a straight-line array, this fixture was not developed for any practical microphone array acoustic measurements such as beamforming, etc. Rather, its primary purpose is to facilitate testing the functionality and connectivity of the PDM 16 module. Nevertheless, it serves as a useful device to demonstrate microphone array testing with the PDM 16 module.

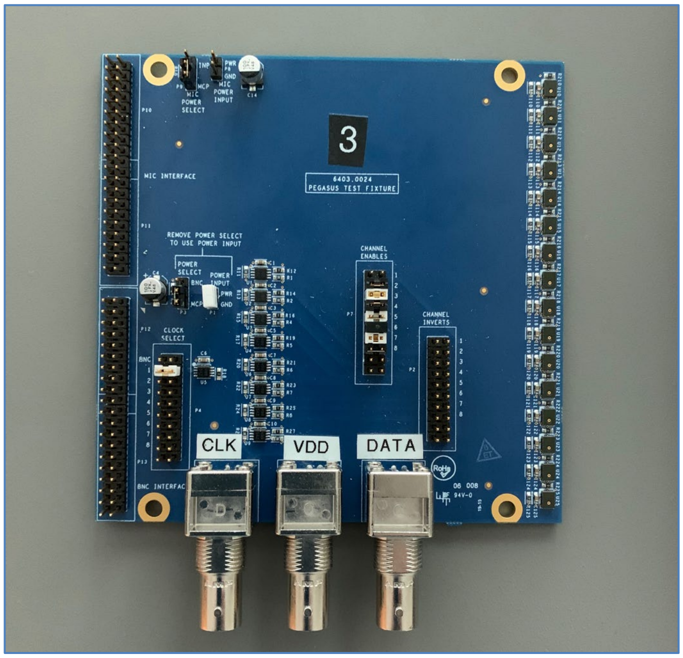

Figure 3. The PDM 16 test fixture with 16-microphone array.

Test Setup

Ideally, microphones should be tested in an anechoic chamber with a low level of background noise, using fixtures that precisely position the device under test (DUT) and reference microphone in the sound field. For this Technote, we used a small test chamber designed for testing hearing aids (Figure 4). This desktop chamber provides approximately 40 dB of noise isolation, is anechoic at 500 Hz and above, and is typically used for acoustic tests in the speech frequency range. It has a full-range loudspeaker that has a flat frequency response within ±3 dB (chamber empty) from 100 Hz to 10 kHz.

Figure 5 shows the microphone array test fixture in the chamber with the linear microphone array positioned on the axis of the loudspeaker. As shown in the photo, a ¼” free field measurement microphone is located next to the second microphone in the array, at the chamber’s “Test Point” – the center of the light blue circle in front of the speaker. The Test Point is a point directly in front of and on-axis to the loudspeaker which is the recommended location for acoustic measurements in this small chamber. The second DUT microphone was positioned at the Test Point because on this fixture, it is PDM channel 1. (For this fixture, from top to bottom in Figure 3, the microphones are wired in the order Ch2, Ch1, Ch4, Ch3, ... Ch16, Ch15.)

Figure 4. Interacoustics TBS25 sound test chamber.

Figure 5. The microphone array test fixture located in the acoustic test box.

Test Equipment

The following test equipment was used for this Technote:

- APx525 Audio Analyzer with PDM 16 module & pod
- APx1701 Transducer Test Interface
- Interacoustics TBS25 Test Chamber
- 378M33 ¼" Free Field Measurement Microphone
- CAL200 Sound Level Calibrator

The APx1701 Transducer Test Interface is a handy accessory for this application, because it has a power amplifier to drive the chamber loudspeaker and constant current power (CCP) for measurement microphones. It also has support for TEDS (Transducer Electronic Data Sheet), meaning the analyzer can automatically calibrate the measurement microphones using calibration data programmed into a chip inside the microphone preamplifier.

Test Procedure

The following test procedure was used to measure the acoustic response of the microphone array.

1. Configure the analyzer for acoustic input (to measure sound pressure from the measurement microphone). Calibrate the mic to measure pressure in Pa and levels in dBSPL (using TEDS, a sound level calibrator, or both).
2. Place the DUT microphone array and the measurement microphone in the test chamber, as shown in Figure 5.
3. Set the analyzer output to Acoustic Mode and use regulation to calibrate the generator in dBSPL at a reference level and at one reference frequency (e.g., 94 dBSPL at 1 kHz).
4. Measure the acoustic frequency response magnitude at the Test Point using the APx Acoustic Response measurement. Repeat the measurement a few times to ensure consistency.
5. Export the Relative Level result from step 4 and use it to equalize the generator output, such that the frequency response magnitude is flat at the Test Point. Check the acoustic output level and if required, repeat the output regulation (step 3).
6. Change the Input Connector to PDM16 and configure the settings as required.
7. Measure the acoustic response of the PDM microphone array using the same settings as step 4.

Leveled Sound Field at the Test Point

Steps 1 to 5 above are used to calibrate the output from the loudspeaker and to equalize it such that the acoustic response is flat at the Test Point. This procedure is described in more detail in Technote 127 – Leveled Acoustic Output.

The APx software has two different methods that can be used to equalize the generator output:

1. Output EQ: Applied in Signal Path Setup, this method uses 30-pole digital filters to equalize all signals generated in a signal path.
2. Measurement EQ: Applied in specific sine-based measurements (including Acoustic Response), this method adjusts the amplitude of the signal at every specified frequency, and can therefore achieve a very flat response.

Figure 6 shows the frequency response magnitude measured at the Test Point in the chamber with no EQ, Output EQ and Measurement EQ. The corresponding deviations from flatness for these three measurements were ±3.33 dB with no EQ, ±0.28 dB with Output EQ and ±0.07 dB with Measurement EQ.

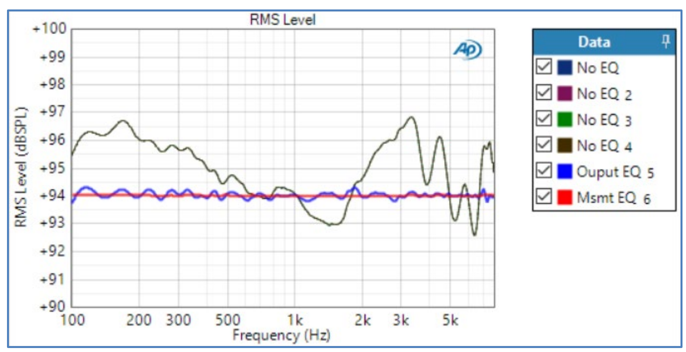

Figure 6. SPL magnitude response at the Test Point with No EQ (4 measurements), Ouput EQ and Measurement EQ.

It should be noted that the above procedure only calibrates the sound field at the Test Point – the response will vary at other locations in the chamber. As, such, with this setup we can only measure the exact response of the DUT microphone at the Test Point (PDM 16 Input Channel 1).

The measured phase response at the Test Point with Measurement EQ is shown in Figure 7. This result (phase – input-to-output excess) represents the phase between the reference microphone and the loudspeaker with propagation delay and any constant group delay removed from the phase calculation.

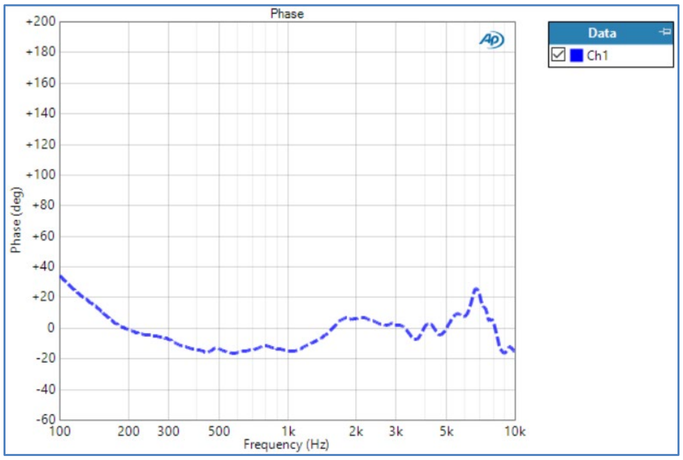

Figure 7. Phase (Input-to-Output Excess) at the test point as measured by the reference microphone.

Configuring PDM 16

Figure 8 shows the PDM 16 configuration settings used for this test. The Input Settings (PDM 16) dialog is invoked by clicking on the gear icon to the right of the Connector control. The settings used are close to the typical values specified on the spec sheet of the DUT's PDM microphones. To measure the PDM 16 inputs with exactly the same outputs as the reference microphone, you can simply change the Input Connector to PDM 16 in the same signal path, or make a copy of the signal path and change the input connector to PDM 16 in the copy.

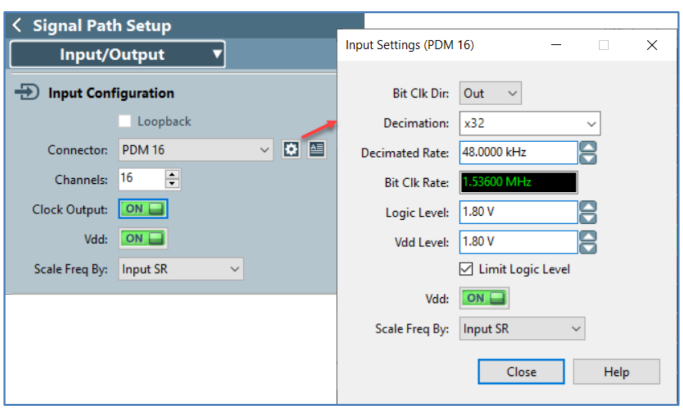

Figure 8. PDM 16 configuration settings used.

Microphone Array Response

Figure 9 shows the magnitude response of the microphone array. Note that of the 16 traces shown, only the blue trace with thick dashed line labeled Ch1 represents the true frequency response of a DUT microphone, because only this microphone was at the Test Point, where the sound field was calibrated.

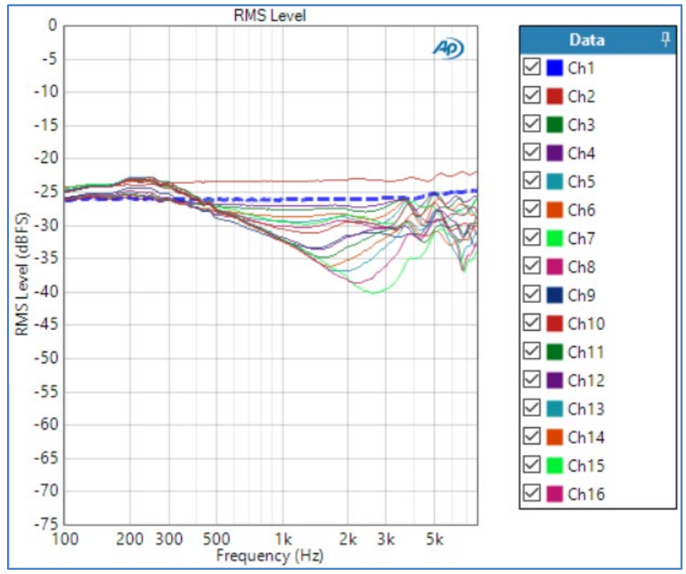

Figure 9. Frequency response magnitude of the 16 PDM microphones in the microphone array.

The shape of the Ch1 trace in Figure 9 is very close to the typical frequency response of this microphone as specified in the datasheet (Figure 10). Note also that for Ch1 the RMS Level at 1 kHz is approximately -26 dBFS. This is, by definition, the sensitivity of the microphone – it's electrical response to a standard acoustic input of 94 dBSPL at 1 kHz. This measured sensitivity is also in agreement with the specified sensitivity for this mic (-26 ±3 dBFS).

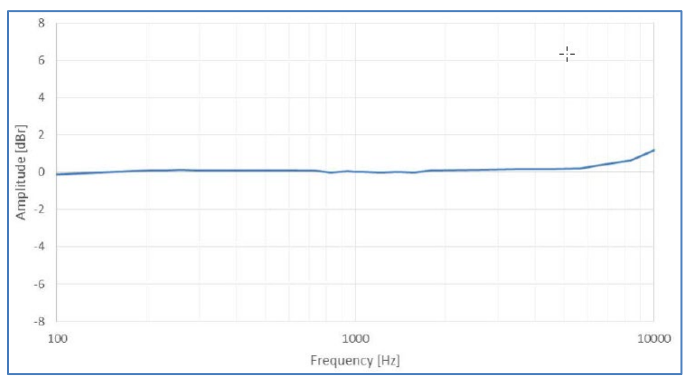

Figure 10. Manufacturer-specified typical frequency response relative to 1 kHz for the microphones in the array.

Figure 11 shows the measured phase response of the microphones in the array. Here again, only the response for microphone Channel 1 is the true phase response. Note that the shape of the Ch1 curve closely resembles the phase measured with the reference microphone (Figure 7).

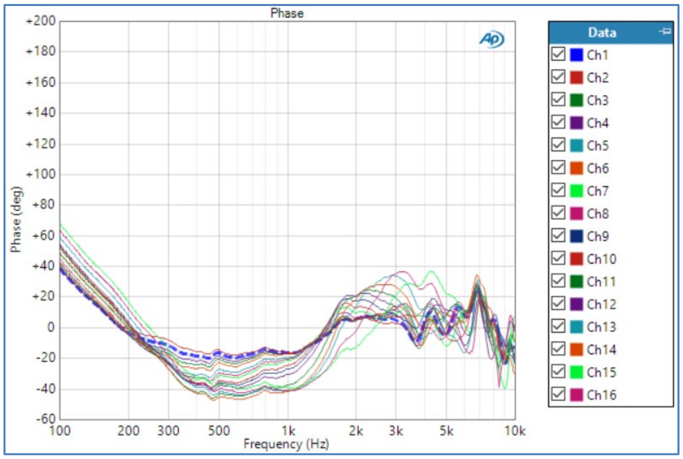

Figure 11. Phase response (input-to-output excess) of the 16 PDM microphones in the microphone array.

Microphone arrays are used for applications like beamforming, to improve the effective signal to noise ratio by combining the inputs from various microphones in the array. Hence the microphone phase response is an important measurement result. Although we can't measure the exact phase response of all the microphones in this linear array in a test setup that has only been characterized at one point, we can get an overall view of the phase relationship between the microphones. The Acoustic Response measurement in APx has a result named Delay. This delay is calculated from the offset of the impulse response, and it represents the average delay between the speaker and the microphone. Figure 12 shows a compare derived result which has been used to calculate the measured delay of each microphone in the array relative to PDM Channel 2, the microphone closest to the speaker.

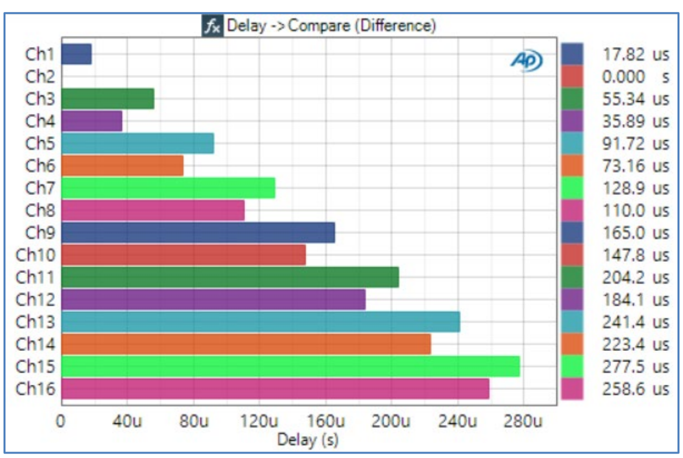

Figure 12. Delay of each microphone in the array relative to microphone position 1 (Ch2), as measured with Acoustic Response.

The pattern is a bit difficult to see in Figure 12, due to the even-then-odd order of the microphones in the array.

However, Figure 13 shows the relative delay data for each microphone plotted against the position of the microphone relative to the first microphone. Note the near perfect linear relationship of these results and that the slope of the line is 0.3503. When converted from mm to meters and µs to seconds, this slope becomes 350 m/s – within about 1% of the estimated speed of sound in air at this temperature and humidity (346 m/s). Not that we're suggesting this as a tool to measure the speed of sound, but it is interesting to see that the physics works out, even over such a small distance.

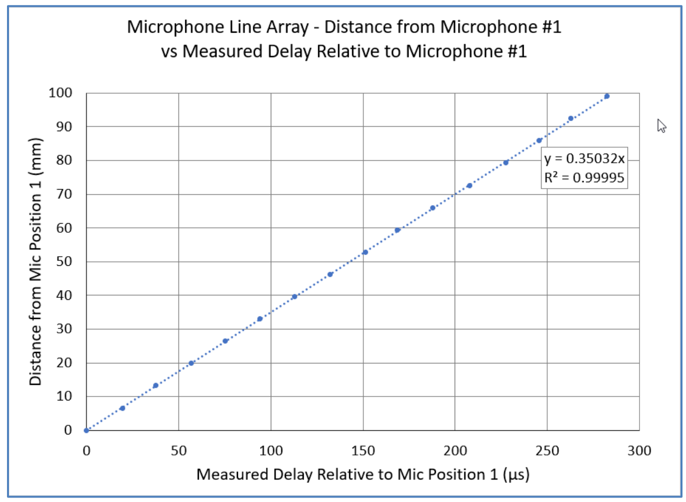

Figure 13. The delay measurements from Figure 12 plotted versus microphone position.

Another interesting way to look at the phase data is in terms of inter-channel phase. The Phase result in the Acoustic Response measurement can also be set to phase relative to Channel 1 (Figure 14). In the case of our fixture, Ch1 is the second closest microphone to the speaker and Ch2 is the closest. The average delay of 18.85 µs between microphones shows up as a phase difference that increases linearly with frequency. At 10 kHz, the phase difference due to one microphone spacing is about 67.9°. This pattern is visible in Figure 13, in which frequency has been plotted on a linear scale. Each curve is predominantly a straight line between 0 degrees at 100 Hz and a multiple of 67.9 degrees at 10 kHz, corresponding to the position of the microphone relative to the Ch1 microphone.

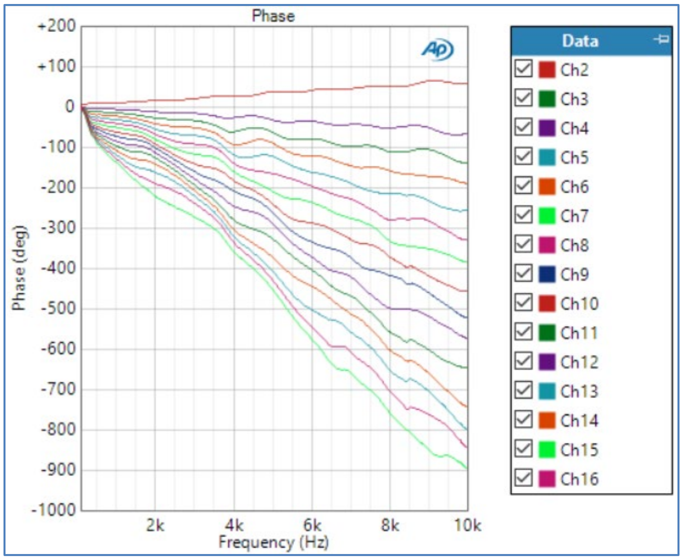

Figure 14. Phase response of the microphones in the array relative to Channel 1 (the 2nd closest mic to the speaker).

Transfer Function Measurements

The APx Transfer Function measurement is also useful for measuring the frequency response of microphone arrays. It allows the measurement of frequency response with any broadband signal, including speech or music. Input signals can also be saved to .wav files as they are acquired and analyzed, which can be quite useful in certain applications. For example, users might want to measure transfer functions with speech signals and then run the acquired signals through a DSP algorithm.

Figure 15 and Figure 16 show the frequency response magnitude and phase, respectively, of our DUT's microphones relative to microphone Ch1 at the Test Point. Note the similarity to Figure 9 and Figure 14.

PDM 16 Applications

An APx audio analyzer with PDM 16 is well suited to research and development of products that use microphone arrays. In addition to a variety of useful audio measurements, the system offers the ability to acquire signals from all microphones simultaneously. These waveforms can be captured as .wav files or exported to MATLAB, which can be useful for evaluating beamforming and speech processing algorithms. And the ability to measure up to 16 PDM microphones simultaneously will be of interest to MEMS microphone manufacturers.

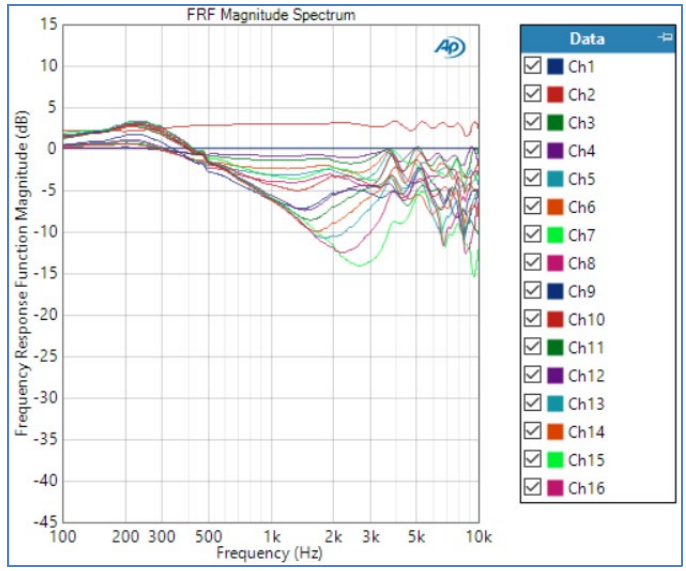

Figure 15. Transfer function magnitude between all DUT microphones and Ch1 microphone (at Test Point).

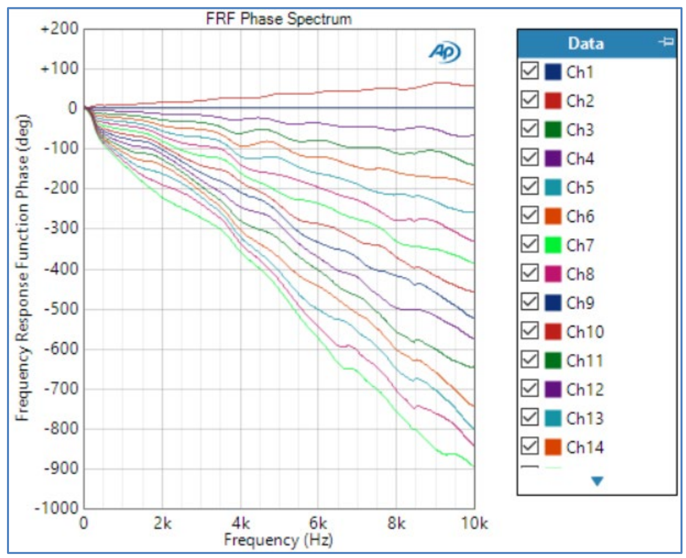

Figure 16. Transfer function phase between all DUT microphones and Ch1 microphone (at Test Point).

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