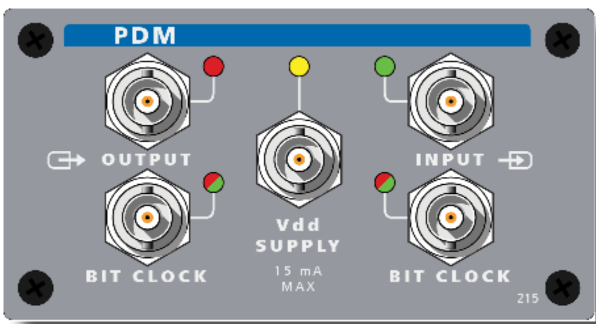
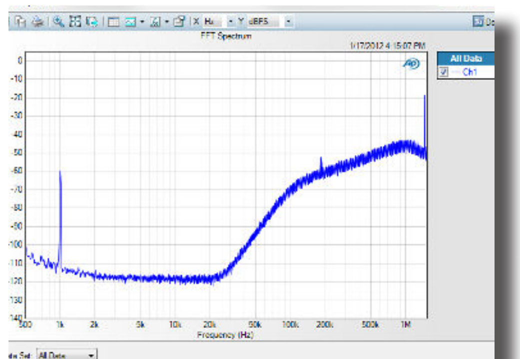
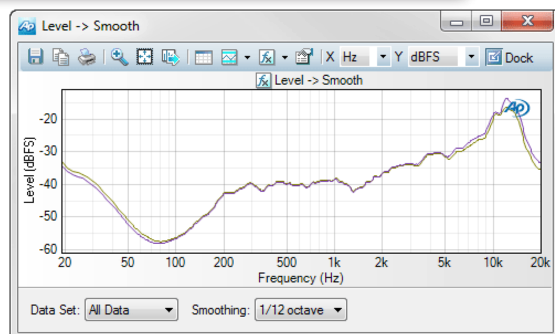
- [ ] 2700 Series
- [x] APx585 family
- [x] APx525 family
- [ ] APx515

Measuring PDM Microphones and Inputs with APx

By Adam Liberman







Introduction

This Technote is intended for those involved with designing and testing PDM devices. We'll focus on microphone components and/or the devices that utilize them, such as smartphones, tablets, and notebook computers, as this is currently the most popular application area for PDM. The information presented, however, is equally applicable to any PDM device, including those using the Sony/Philips DSD format.

We'll first review how PDM works, describe relevant audio tests, look at the APx PDM module, and conclude with hands-on testing of a PDM MEMS microphone and a PDM microphone input.

What is PDM?

PDM stands for pulse density modulation. However, it is really better summarized as "Oversampled 1-bit audio." It's a high sampling rate, 1-bit digital system. If you increased the sample rate of CDs by a factor of 64 (referred to as "interpolation"), and reduced the wordlength from 16 bits to 1 in a reasonable way, you'd have the basis of a PDM system.

An undithered 16-bit system has a theoretical signal-to-noise ratio of around 98 dB. An undithered 1-bit system has a signal-to-noise ratio of about 8 dB. However, by use of noise shaping, most of the noise is pushed above the audible band (above 20 kHz) and performance can be very high.

PDM has become a popular format for transmission of audio from the microphone to the signal processor in smartphones and other consumer devices, due to its low cost, immunity from interference, and simplicity of mic integration using MEMS technology.

For an in-depth discussion of PDM technology, read the paper "Understanding PDM Digital Audio," available for download at ap.com.

APx PDM Module

Figure 1: APx PDM module.

AP's PDM interface module is available on all APx analyzers except the APx515. It features both PDM inputs and outputs, for testing PDM MEMS microphones as well as PDM inputs on chipsets.

The APx PDM module includes the following features:

- UI and programmatic control over physical interface parameters: logic and V_{DD} voltages, sample rate, data edge, clock direction, modulation order (4th or 5th), interpolation (32–800x), and decimation (1–800x).
- Analysis is in the digital domain to achieve high accuracy (as opposed to analysis after analog conversion). FFTs may be viewed using the raw PDM bitstream or the decimated audio—the PDM bitstream mode includes the PDM spectrum above the audible audio band.
- Single or swept frequency Power Supply Rejection measurements, utilizing the PDM module's built-in V_{DD} power supply output.
- Cross domain / cross interface measurements, combining the PDM input or output with any of the following analyzer outputs or inputs: analog balanced, analog unbalanced, digital balanced (AES3), digital unbalanced (SPDIF), digital optical (TOSLINK), serial digital, HDMI, or Bluetooth (some interfaces are options and not present on all instruments). PDM output with PDM input may also be configured.
- Comprehensive set of audio measurements.

Setting Up Tests of PDM Microphones

Electrical connections:

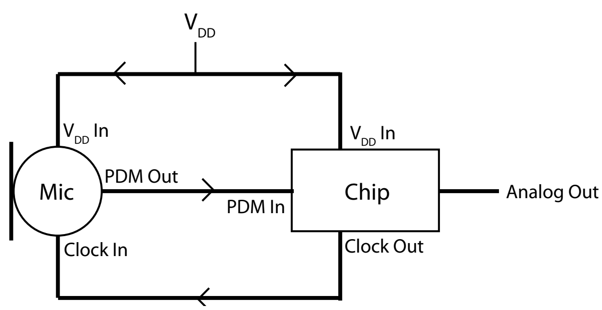


Figure 2: Typical PDM mic to chip connections.

Open APx500 and configure the signal path. Typically the Output Configuration will be set for Analog Unbalanced or Analog Balanced, to feed a self-contained powered speaker, a power amplifier and speaker, or an amplifier and mouth simulator. Connect the analyzer outputs to the amplifier or speaker.

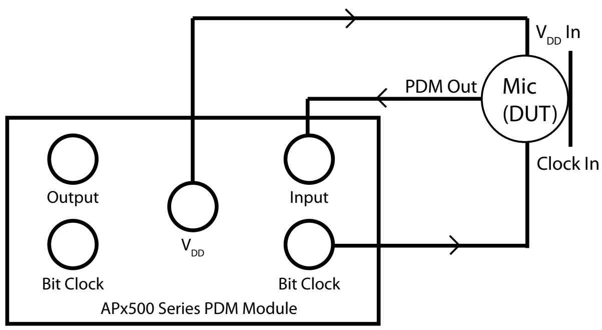


Figure 3: Mic to APx PDM connections.

Set the Input Configuration to PDM. Click **Settings** to open the *Input Settings* dialog. The default settings are correct for most microphones; however, if necessary, you can change the bit clock direction, decimation ratio, decimated rate, Data Edge, and Coupling. Also, make sure that the V_{DD} Level and Logic Level are set appropriately for the device.

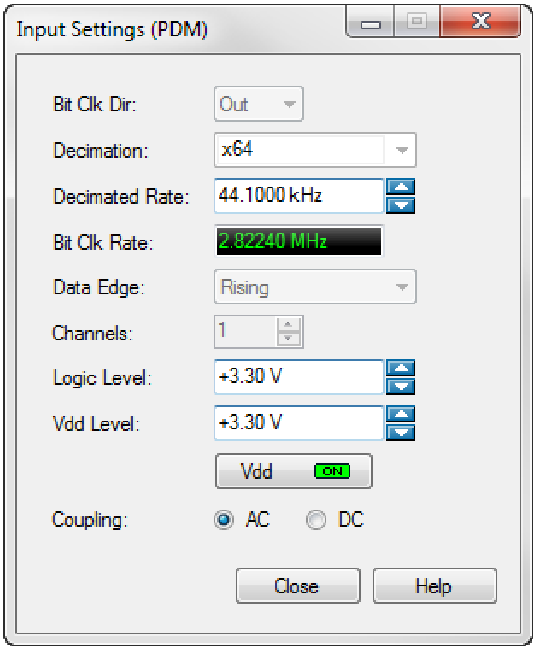


Figure 4: PDM Input Settings window. The settings shown here are appropriate for the Sony/Philips DSD format.

Now, connect the microphone's data, clock, and V_{DD} to the PDM module on the analyzer. If the cable length to the PDM microphone is more than 1 meter, it will be necessary to insert the AP PDM Line Driver accessory in line near the microphone, as most PDM microphones don't have enough current to drive long cables. AP's PDM Line Driver can drive cables up to 150 feet, depending on clock rate (45 feet at 3.072 MHz). This makes it possible to locate the mic in a quiet room or acoustic chamber.

Figure 5: AP PDM Line Driver.

Click the V_{DD} button on the Input Settings dialog or in Signal Path Setup to supply power to the microphone.

Figure 6: APx PDM module, connecting the output of a PDM MEMS microphone to the PDM input on the APx analyzer.

Acoustic Configuration

Acoustic measurements will require an acoustic sound source; a testing chamber, room, or simulator; and hardware to hold the mic in a fixed position.

Figure 7: KEMAR 45MB HATS manikin with mouth simulator.

When testing in a room or acoustically treated chamber or box, the sound source is normally a small but capable speaker with a built-in or outboard amplifier. The speaker needs to be high quality, having relatively flat response across the audible bandwidth (20 Hz to 20 kHz), low distortion, and a high overload level. Limiting the size helps the speaker to approximate a point source. For IMD measurements, a second speaker is necessary.

Testing in a room or chamber is normally done to provide a look at basic device performance without the influence of reflected sound. However, in actual usage, a PDM microphone will typically be mounted in a device like a smartphone or tablet which is held close to the mouth. It is therefore just as important to measure performance under these circumstances, as they can greatly affect the results, especially when measuring frequency response, directivity, and phase. To simulate this configuration in a repeatable and standards-defined manner, a head and torso simulator is used as the signal source, and the phone is held in the proper place using a handset positioner.

Calibrating Acoustic Level in dBSPL

In order to make the measurements, we need to calibrate the analyzer to display voltage in equivalent dBSPL. We'll do that using the following process:

1. Calibrate the analyzer to read the signal from a measurement microphone in dBSPL (using a pre-calibrated microphone calibrator).

2. Remove the calibrator and adjust the analyzer output level until the stimulus speaker puts out 94 dBSPL, as measured using the same microphone.

3. Substitute the microphone under test for the measurement microphone, switch the analyzer Input connection to PDM, and drive the stimulus speaker at 94 dBSPL, using the analyzer output level determined in the previous step.

4. We know that the microphone under test is now receiving a level of 94 dBSPL from the speaker, so we can now recalibrate the analyzer so that the level it is currently measuring (in dBFS) is equal to 94 dBSPL1.

If you do not have a calibrator, or prefer to enter the microphone sensitivity manually, you can skip step one and type millivolts per Pascal (mv/Pa) directly into the dBSPL1 and dBSPL2 boxes (Calibrator Level boxes should be set to 94 dBSPL). Now, let's actually perform the calibration:

First, connect a measurement microphone to its power supply, and then connect the audio output of the power supply to the APx analyzer. Typically, a free-field capsule will be used for room measurements, and a pressure capsule will be used for the KEMAR head and torso simulator or for an enclosed box.

Figure 8: Mic and pistonphone calibrator.

Place an acoustic calibrator over the measurement microphone and turn it on. We now need to measure the voltage level on the analyzer inputs and reference it to the calibrator level. In APx500, this can be conveniently done by going to *Reference Levels* in the navigator tree and then clicking **Set dBSPL**. In the calibration dialog that appears, make sure that the *Calibrator Level* field value matches the output level of the calibrator device being used (94 dBSPL is most common, but some are 74, 104, or 114 dBSPL), and then click on Set 1 (for channel 1). This will set the currently measured voltage equal to the sound pressure level of the calibrator.

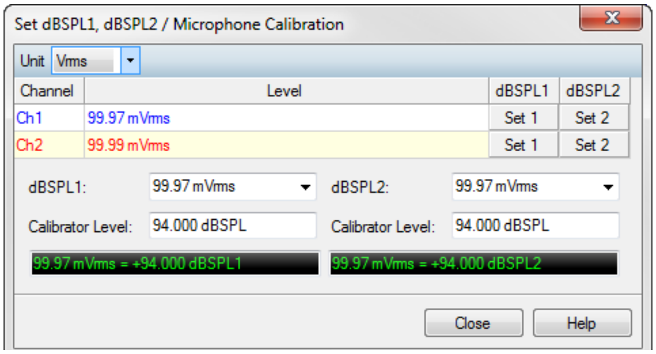


Figure 9: Set dBSPL dialog.

Now, remove the calibrator and place the measurement microphone in the position where the PDM microphone under test will ultimately be placed. The KEMAR 45BM head and torso simulator (HATS) comes with a calibration jig to hold a 1/4" measurement microphone in the correct spot.

Turn on the analyzer output to drive the stimulus speaker. In the case of the KEMAR mouth simulator, the speaker is built inside the mouth. In *Reference Levels*, click the **AutoGen Level** button. In the dialog that opens, set target *Measured Level* to 94 dB SPL1 and then click **Set Generator Level**. When this has been achieved, click **Set From Level**. This will calibrate the generator reference (0 dBrG) to equal the voltage level needed to produce 94 dBSPL from the stimulus speaker. Once this has been calibrated, make sure not to change any gain settings on the amplifier driving the speaker.

Change *Input Configuration* back to PDM, remove the measurement microphone, and then place the PDM microphone under test into the box in the same location.

Turn on the APx generator at a level of 0 dBrG to drive the speaker at 94 dBSPL. Click on **Set dBSPL** again to open the *Microphone Calibration* dialog. Click on **Set 1** (for channel 1) to recalibrate the analyzer input to read dBSPL correctly with the PDM microphone.

When using the KEMAR 45BM head and torso simulator (with its built-in mouth simulator), please refer to its documentation for additional details about the level and response calibration procedure.

Setting Up Tests of PDM Inputs

Open APx500 and configure the signal path. Set the Output Configuration to PDM. Click Settings to open the Output Settings dialog. The default settings are correct for most PDM device inputs; however, if necessary, you can change the bit clock direction, interpolation ratio, modulator order, data edge, and the frequency scaling reference.

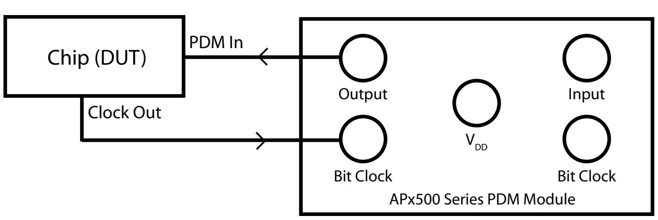


Figure 10: Connecting the APx PDM module output to the PDM input on a device.

Now, connect the device's data and clock to the PDM module on the analyzer.

Set the Logic Level appropriately, as this will affect the level of the data being sent out, as well as the voltage threshold of the bit clock receiver (assuming Clock Direction is set to In). Normally, it is not necessary to set V_{DD} Level or turn on V_{DD}, as chips with a PDM input will usually have their own power supply. If utilizing the analyzer's V_{DD} supply, keep in mind that it is primarily designed to power small PDM microphones and has a maximum output of 15 mA.

Figure 11: APx PDM module, connecting the APx PDM output to the PDM input on a device.

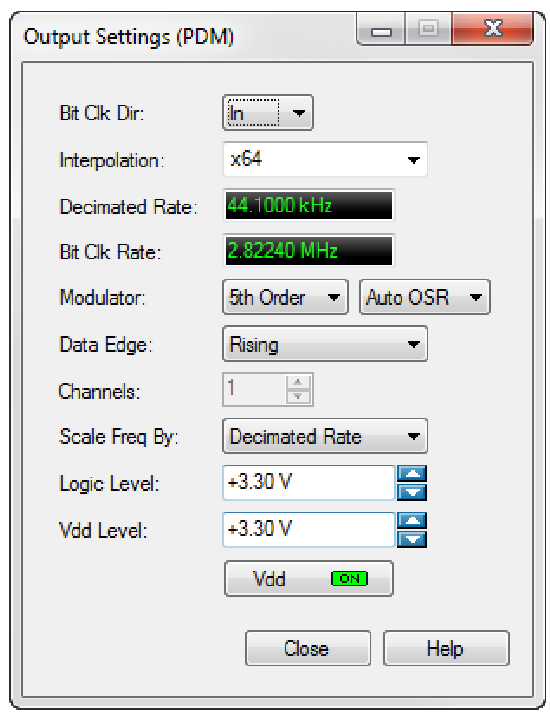


Figure 12: PDM Output Settings window. The settings shown here are appropriate for the Sony/Philips DSD format.

PDM Measurements

PDM audio measurements are generally similar to those for other audio devices. However, there are some details particular to PDM testing.

The six basic audio measurements (level, frequency response, THD+N, phase, crosstalk, and signal to noise ratio) form a good framework for discussing comprehensive PDM device testing (see *Technote 104: Introduction to the Six Basic Audio Measurements* available at ap.com). Most other audio measurements are variations on these. In the following, we'll discuss each one as they relate to both PDM microphone and PDM input testing.

Measuring PDM Microphones

Level:

Level measurements include overload level and maximum acoustic input level at 1 kHz. Overload is normally specified as the point where THD+N or THD equals 3%. In traditional microphone measurements, overload and maximum input are synonymous. However, due to the distortion characteristics of PDM microphones, a second measurement is often made to determine the maximum acoustic input level that is needed to produce full output from the mic, without regard to distortion level.

Overload and maximum acoustic input level can be determined using the Stepped Level Sweep measurement, after following the dBSPL calibration procedure described above. To make sure that the source speaker can produce the required sound level for the test, first run the test using a calibrated measurement microphone. Measurement microphones can usually tolerate 145–160 dBSPL (check the datasheet for your capsule), which is well over the level that most PDM microphones can accept.

The APx Stepped Level Sweep includes results for level, linearity, and THD+N—all of which can be used to determine the overload and maximum levels. The Maximum Output measurement may also be used to automatically determine the 3% overload level.

Level measurement will also be discussed in the context of other measurements below.

Frequency Response:

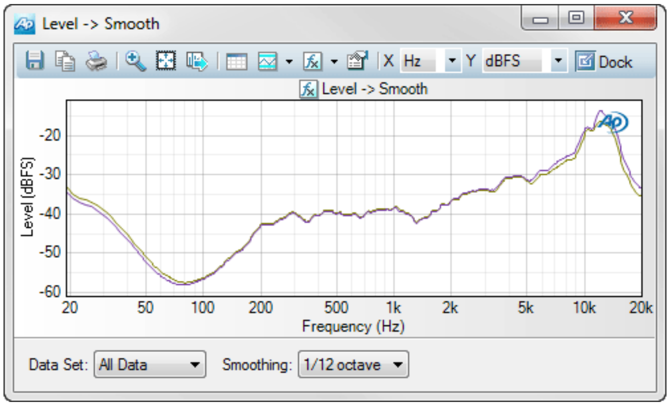


Figure 13: Acoustic Response measurement, Level result, with smoothing applied.

Frequency response is plotted using the Acoustic Response measurement. In this measurement, you may adjust the time window cursor in the Energy Time Curve or Impulse Response results to remove the influence of room reflections. More details about how to do this are included in APx500's built-in help. The frequency response graph below uses the APx500 derived result *Smoothing*, set to 1/12 octave, to smooth out small response variations.

Distortion:

The standard THD+N measurement usually can not be used to assess microphone distortion, because the distortion of the stimulus loudspeaker itself will typically dominate and obscure distortion caused by the microphone. The THD+N measurement can be useful when assessing the 3% overload and maximum output points, if done using a source speaker that has been verified to have very low distortion.

To measure distortion without the sound source influencing the results, we can measure IMD (intermodulation distortion) using two stimulus speakers. IMD uses a dual tone signal and measures distortion products created when the two stimulus frequencies interact with each other, or intermodulate. By playing each tone through a different speaker, we ensure that no IMD distortion will be generated by the speakers, and that any that we measure is caused within the microphone itself.

The speakers will generate harmonic distortion, but the harmonics will not coincide with the intermodulation products and therefore will not affect the measurement.

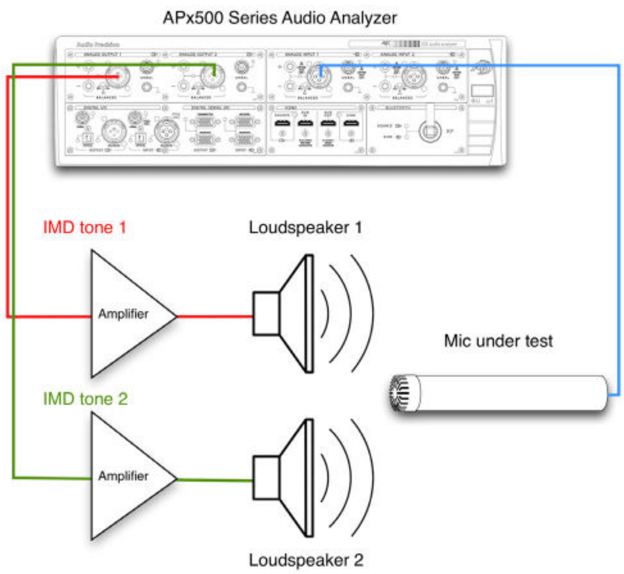


Figure 14: IMD measurement using split channels.

To enable IMD measurements in APx500 with the two tones on different channels, check the "split" checkbox in the Generator Settings panel (requires APx500 v3.1 or later).

IMD should be measured at various levels, including typical voice levels, the standard 94 dBSPL, and at higher levels.

Noise:

Signal-to-noise ratio for microphones is often specified as the difference in output level between the microphone's self-noise and the output produced with a standard 94 dBSPL acoustic signal. Similarly, dynamic range is specified as the difference between the self-noise level and the mic's maximum output level.

The 94 dBSPL output level has already been determined during the acoustic calibration procedure above, and the maximum output has been determined with the Stepped Level Sweep. The only one that remains to be measured is the self-noise.

In order to make the measurement, you will need to isolate the microphone from acoustic noise by placing it in a very quiet room, and furthermore usually inside an isolation box within the room. The ventilation system should be turned off to prevent noise and rumble, and noise sources like the analyzer and computer should not be located close by. MEMS mics typically have a self-noise level of around 30–35 dBSPL. A very quiet room is also typically in the range of 30–35 dBSPL, so usually the box will be a necessity to provide additional isolation. Plugging the sound port on the mic may help reduce noise pickup.

You should first make the noise measurement using a measurement microphone to verify that the noise level in the room is low enough to accurately characterize the PDM microphone. Check the specifications for your measurement microphone to make sure that it is suitable. Most general purpose 1/2" free-field capsule have self-noise between 15–20 dBSPL.

Once the microphone has been isolated, go to the APx500 Noise (RMS) measurement and set the graph to read in dBSPL. The 20 kHz low-pass and 20 Hz high-pass filters should be selected. Set the weighting filter to A-wt if your specifications require A weighting. Now read the measurement on the graph.

Phase:

Although most PDM microphones are mono, some stereo and noise cancelation modules are available. Relative phase may be measured on these devices using the Interchannel Phase measurement. The Acoustic Response measurement can also be used, and includes results for phase, group delay vs. frequency, and delay.

Crosstalk:

Crosstalk isn't usually relevant for acoustic measurements, since it would be difficult to acoustically isolate each half of a stereo device.

PSR:

Power supply rejection is of particular interest for PDM mics, due to the electrically noisy environment in which they normally operate. Because the APx PDM module includes a controllable power supply, measurement of PSR—especially swept frequency PSR—is very convenient to do.

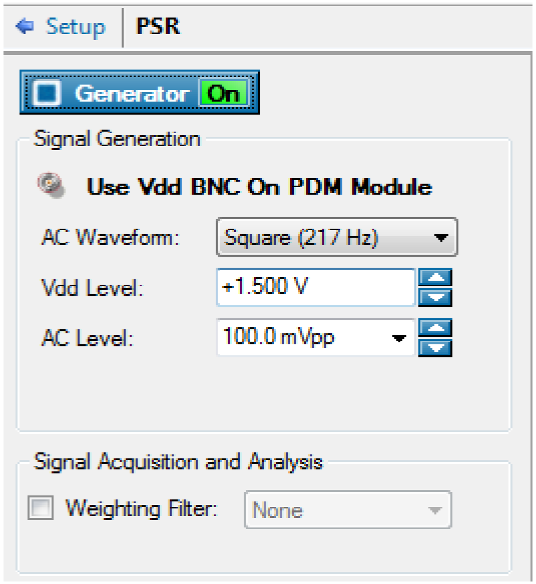


Figure 15: PSR measurement.

The PSR of PDM mics is determined by superimposing an AC wave onto the DC power supply, and then measuring the level of the AC wave that is present in the audio. The PSR measurement has a selection of waveforms, including sine wave, 217 Hz square wave, pulsed 217 Hz square wave (duty cycles from 1/8 to 7/8), and TPDF noise. PSR is commonly specified on datasheets using a 217 Hz square wave at 100 mVpp. It's used because GSM phones have a 217 Hz switching frequency that is the main source of noise on the power supply.

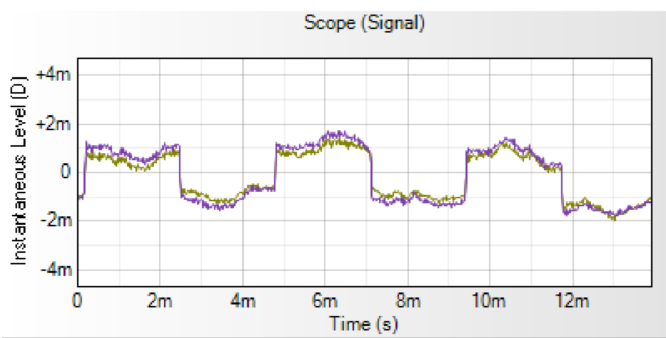


Figure 16: APx500 Scope monitor showing the 217 Hz square wave leaking into the audio signal.

Bandpass filtering is applied to the audio before measurement. In the case of the sine wave, the filter is at the stimulus fundamental; for the square wave, it is at the fundamental and odd harmonics (to H_{101}); and for the pulsed square wave, it is at the fundamental and all harmonics (to H_{101}). The filtering prevents random noise outside the frequencies of interest from affecting the results. It also gives the measurement some immunity to room noise being picked up by the mic, but for the most accurate results, the microphone should be isolated from noise using a chamber or box. No bandpass filtering is applied when using the TPDF noise stimulus, so it is especially important in this case to isolate the microphone.

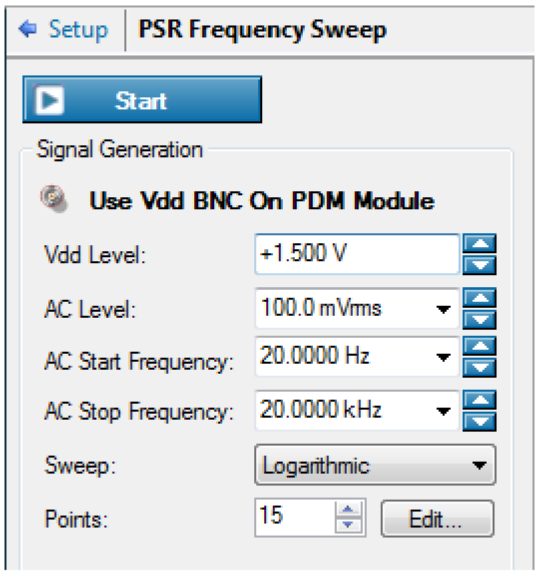


Figure 17: PSR Frequency Sweep measurement.

PSR Sweep:

This measurement is similar to PSR, but the AC riding on the DC is swept, and only a sine wave is used. The microphone should be isolated from outside noise for accurate readings.

V_{DD} Ramp

V_{DD} Ramp is a unique APx500 measurement that ramps the V_{DD} voltage from low to high over a predetermined length of time. The generator output is off during this time.The analyzer input is plotted vs. time to detect any disturbances or noise on the device output.

Signal Analyzer:

The APx500 Signal Analyzer provides an FFT spectrum, as well as Amplitude Spectral Density, Power Spectral Density, and a scope view. Unique to APx500 is the ability to view the PDM spectrum before decimation, allowing analysis of the noise spectrum up to the sampling rate of the device. This is done by setting the Signal control in Signal Analyzer to PDM Bitstream.

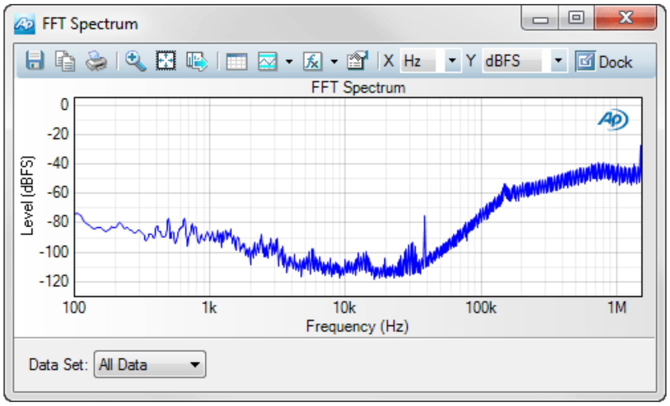


Figure 18: APx Signal Analyzer, FFT Spectrum view of PDM Bitstream.

Measuring PDM Inputs

Measuring a device with a PDM input is similar to measuring a device with an SPDIF digital unbalanced input, with a few differences noted below.

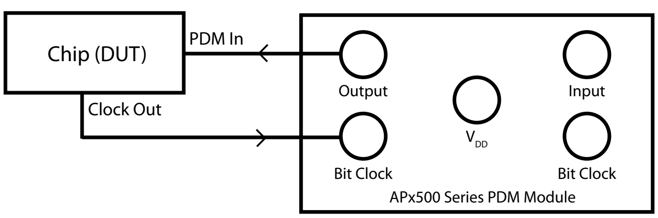


Figure 19: Connecting the APx PDM output to the PDM input of a device.

Frequency Response:

Frequency response may be measured using the Frequency Response or Continuous Sweep measurements. The Continuous Sweep measurement is like the Frequency Response measurement, but gives additional results.

Distortion:

THD+N and THD results are available from a number of measurements. The Stepped Frequency Sweep measurement gives both THD+N and THD across the frequency spectrum. Additionally, the Continuous Sweep measurement gives THD results, the Multitone Analyzer measurement gives TD+N, and the THD+N measurement gives single frequency results that include THD+N, THD, Noise, and distortion products. IMD may be measured using the IMD, IMD Frequency Sweep, and IMD Level Sweep measurements. To determine dynamic range (in conjunction with the noise measurement below), use the THD+N, Stepped Level Sweep, or Maximum Output measurements to find the 3% THD+N point. Note that the distortion in the analyzer's PDM modulator rises when it starts to overload, beginning around -10 to -8 dBFS, depending on settings. This is a characteristic of PDM and is not a deficiency in the analyzer. When the overload point is reached, an overload indicator will flash in the APx500 status bar. The point where the indicator just begins to flash occasionally will typically be the lowest distortion.

Phase:

If you are testing the PDM input in stereo mode, you can measure relative phase using the Interchannel Phase or Continuous Sweep measurements.

Absolute delay and group delay results are available in Acoustic Response, and group delay is available in Continuous Sweep.

PSR:

The PSR and PSR Sweep measurements may be used to evaluate devices with PDM inputs. More details about the PSR and PSR Sweep measurements are in the section above on testing PDM microphones. Note that the APx PDM module V_{DD} supply is limited to 15 mA and is primarily intended to power PDM microphones, which typically draw about 1 mA. If the device you are powering demands additional current, you will need to connect a unity gain power amplifier between the PDM module's V_{DD} output and the device's power input.

Noise:

Use the Noise (RMS) measurement to determine the noise floor in dBFS.

Testing Control Codes

Some PDM devices can send or receive control codes. A control code is a 2-digit hex number (8 bits) that is inserted into the PDM bit stream, replacing the audio for the duration of the control code sequence. The function of the codes varies by device. Well-behaved devices will prevent the codes from being played as audio.

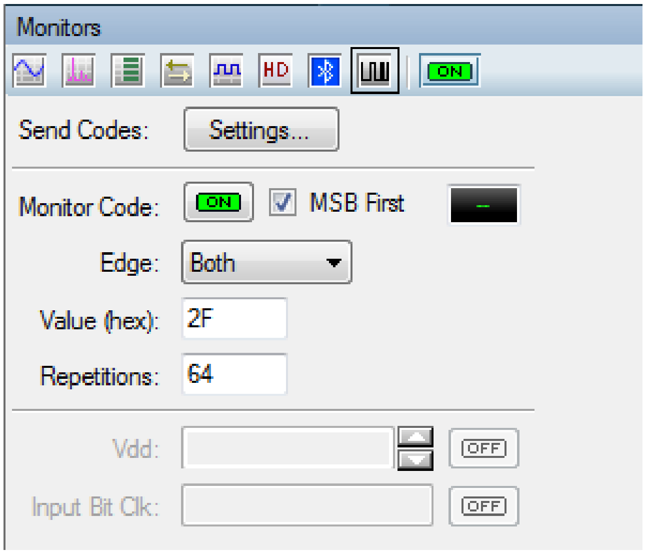


Figure 18: PDM Monitor.

The PDM Monitor (grouped with the other Signal Monitors) lets you monitor for specific incoming codes and repetition counts on the PDM Input (the status bar also indicates this). It also lets you open a dialog to send codes on the analyzer's PDM Output. You may also send PDM codes by adding sequence steps in the APx500 Navigator.

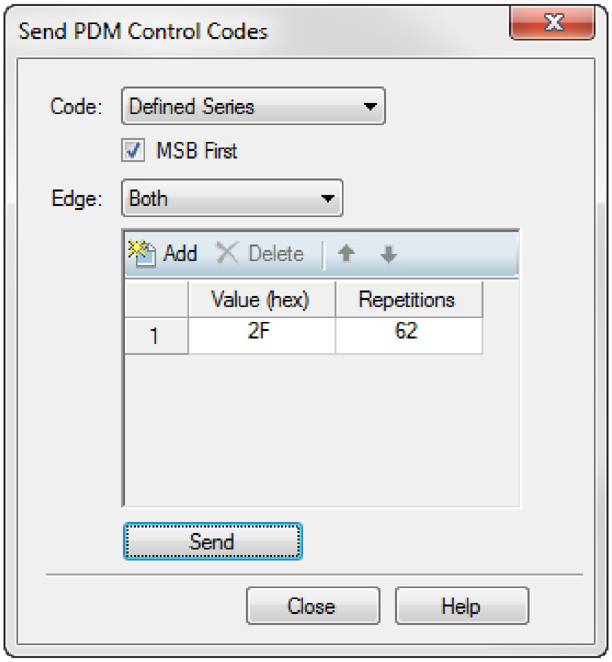


Figure 19: Send PDM Control Codes.

Resources (links):

TN117: Measuring PDM Microphones and Inputs with APx (this Technote)
http://ap.com/display/file/624

Understanding PDM Digital Audio
http://ap.com/display/file/612

APx PDM Option (web page)
http://ap.com/products/apx/pdm

APx PDM Option Technical Details
http://ap.com/display/file/619

APx PDM Option Specifications
http://ap.com/display/file/656

PDM Specs

Sample rate range
4 kHz to 216 kHz

Bit clock range
128 kHz to 24.576 MHz

Oversampling ratio
32–800x

Decimation ratio
1x–800x

Interface logic levels
1.8–3.6 VDC

Edge modes
Rising, Falling, Stereo (LR), Stereo (RL

Modulator
4th or 5th order

Modulator maximum input level
0 dBFS

SNR
127 dB (1 kHz, .45 FS BW, unwtd, 256x oversampling, 5th order modulator)

THD+N
–128 dB (1 kHz, –9.8 dBFS, 20 kHz BW, unwtd, 256x oversampling, 5th order modulator)

Dynamic Range
137 dB (AES17, CCIR-RMS, 256x oversampling, 5th order modulator)

Flatness
± 0.001 dB (20 Hz to 20 kHz)

DC output
1.5–3.6V, 15 mA max

Connectors
Output data, output clock, input data, input clock, external power (all via BNC)

APx500 Series Audio Analyzer Key Specifications

SYSTEM PERFORMANCE

Residual THD+N (20 kHz BW)
–105 dB + 1.3 μV [APx525 family]
–103 dB + 1.4 μV [APx585 family]

GENERATOR PERFORMANCE

Sine Frequency Range
0.1 Hz to 80.1 kHz [APx525 family]
5 Hz to 80.1 kHz [APx585 family]

Frequency Accuracy
2 ppm [APx525 family]
3 ppm [APx585 family]

IMD Test Signals
SMPTE, MOD, DFD

Maximum Amplitude (balanced)
21.21 Vrms [APx525 family]
14.4 Vrms [APx585 family]

Amplitude Accuracy
±0.05 dB

Flatness (20 Hz–20 kHz)
±0.008 dB

Analog Output Configurations
unbalanced & balanced

Digital Output Sampling Rate
22 kHz–192 kHz

Dolby / DTS Generator
Yes

ANALYZER PERFORMANCE

Maximum Rated Input Voltage
300 Vrms (bal) / 160 Vrms (unbal) [APx525 family]
110 Vrms (bal/unbal) [APx585 family]

Maximum Bandwidth
>90 kHz

IMD Measurement Capability
SMPTE, MOD, DFD

Amplitude Accuracy (1 kHz)
±0.05 dB

Amplitude Flatness (20 Hz–20 kHz)
±0.008 dB

Residual Input Noise (20 kHz BW)
1.3 μV

Individual Harmonic Analyzer
d2–d10

Max FFT Length
1.2M points

DC Voltage Measurement
Yes

ACCREDITED

Accredited by A2LA under ISO/IEC: 17025 for equipment calibration

Specifications subject to change.

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