



QUALIFYING AN AUDIO INTERFACE FOR APx500 FLEX

by Joe Begin

About This Technote

In 2019, Audio Precision announced the release of APx500 Flex, a software license that enables the APx500 software to be used for audio test without Audio Precision hardware. With APx500 Flex and a third party ASIO¹ driver-enabled audio interface (sound card), users have access to the extensive suite of audio measurements available in the APx500 software at a much lower price point.

Although an audio interface with the right software can be used successfully for audio test, these devices are primarily designed for recording and mixing music. As such, choosing an appropriate audio interface and configuring it for precision audio measurements can require a significant amount of work. To make it easy for users, we've already done this work and verified the compatibility of three different audio interfaces for use with APx500 Flex. Instructions and configuration files for these three compatible audio interfaces are available on www.ap.com.

We've also anticipated that some users will want to use an unsupported audio interface. In this Technote we discuss the process of qualifying an audio interface for audio test with APx500 Flex. We also cover some of the potential pitfalls of using an audio interface, rather than purpose-built test hardware, for audio measurement.

Note: APx500 Flex only works with audio interfaces that have an ASIO™ driver.

About APx500 Flex

The APx500 software has had support for ASIO devices since version 3.1, introduced in 2012. Since then, users

have been able to select an ASIO device as the APx Output Connector, Input Connector, or both. Although this capability has been an extremely useful addition to the APx feature set, prior to APx500 Flex and the 5.0.2 release, it had two significant constraints:

1. A physical APx500 audio analyzer was required to be connected to the PC for the software to even run.
2. ASIO devices were handled like any other digital IO device, with digital units (D or FS) being assigned to both inputs and outputs. This makes it inconvenient to generate or analyze analog voltage signals using an ASIO device.

With APx500 Flex (supported as of APx v5.0.2), these constraints have been removed. ASIO devices can now be selected as both the Output and Input Connector when an APx500 Flex Key is connected: no other physical APx hardware is required. And ASIO I/O channels can now optionally be scaled automatically to volts (and subsequently to pascals/dBSPL for acoustic measurements).

Figure 1 shows the Scaling Mode and Scaling Factor controls for ASIO outputs and inputs added in APx500 v5.0.2. Each scaling mode control has selections of Digital and Analog. When Digital is selected, the signal units are D or FS, as they have been since ASIO support was first introduced in 2012. When Analog is selected, the Scaling Factor control becomes visible. This is used to assign the factor needed to scale the output or input signal from digital to voltage units (i.e., to “calibrate” the output or input). As shown, there is one output scaling factor for all channels,

latency, high performance and stability, ASIO has become a supported standard among leading audio device manufacturers in the form of ASIO drivers for their products.

¹ ASIO™ stands for Audio Stream Input Output, a protocol developed by Steinberg in 1997 for direct communication between a Windows software application and a PC sound card. With its low

while on the input side, there is a separate scaling factor for each channel.

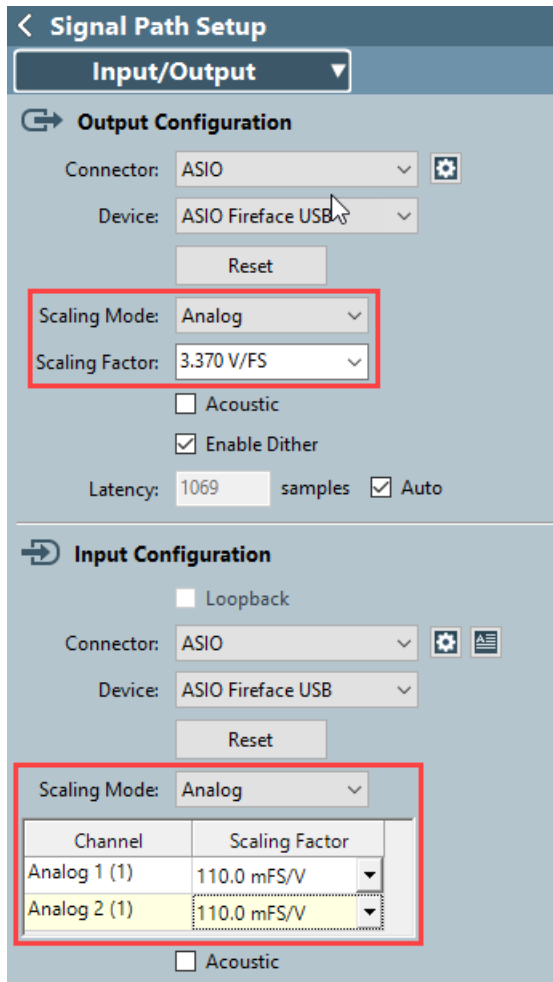


Figure 1. ASIO Scaling Mode and Scaling Factor controls in APx version 5.0.2

Basic Calibration vs Qualifying an Audio Interface

A basic calibration of an audio interface is required before it can be used to conduct audio measurements of devices with analog (or acoustic) I/O. Basic calibration can be performed using a few simple steps with a precision AC volt meter (or an audio analyzer). The procedure involves measuring the output and input scaling factors and assigning them to the corresponding controls in Signal Path Setup of the APx500 software.

While basic I/O calibration of an audio interface is sufficient to get you started at making analog audio measurements, to truly have confidence in the measurements, the audio interface should be qualified. This involves a more rigorous and in-depth procedure using an audio analyzer to measure

key input and output performance attributes of the audio interface, including:

- Sensitivity (or scaling factors)
- Frequency Response
- Residual distortion
- Noise
- Measurement range
- Delay compensation and stability
- Presence of glitches and dropouts

Audio Interface Connections

A major consideration when using an audio interface for audio test is using the right connections to the device under test (DUT). Both the physical connector and the connection type (balanced or unbalanced) are important.

Physical Connector Type

For many years, the professional audio industry has used the 3-pin (AES-14 standard) XLR connector (Figure 2, left) for balanced analog connections. These connectors use the standard convention that pin 1 is connected to the cable shield, while pins 2 and 3 carry the positive and negative poles of the differential signal.



Figure 2. Standard XLR audio connectors (left) and 1/4" TR and TRS phone plugs (right).

More recently, professional audio devices have begun using 1/4" tip-ring-sleeve (TRS) phone plugs (Figure 2, lower right) for balanced connections, no doubt because they take up less space on I/O panels. One potential pitfall of using 1/4" phone plugs is that the 1/4" phone jack that they mate with will also accept 1/4" tip-sleeve (TS) phone plugs (Figure 2, upper right), which can only be used for unbalanced connections. If a TS phone plug is inserted into a TRS jack, it will ground one side of the balanced connection, potentially

reducing the expected voltage dramatically¹. Also, it's impossible to tell from the outside whether a ¼" phone jack is TRS or TS, unless it is labeled as such.

Some audio interfaces have a third type of connector, the XLR combo jack, as shown for the connectors labeled 1 and 2 in Figure 3. These connectors will accept either a standard XLR male connector or a ¼" phone plug. They are often used on one or two input channels of an audio interface that provide a microphone pre-amplifier circuit. Users take heed! Typically, the input characteristics of these channels are completely different depending on whether an XLR connector or ¼" phone plug is inserted. For example, the XLR function of the connector usually supports switchable phantom power for condenser microphones, uses a different gain stage and has a different input impedance than the ¼" phone jack part of the connector.



Figure 3. A portion of the front panel of an RME Fireface UC audio interface showing XLR combo jacks (1-2) and TRS jacks (3-4)

Balanced vs Unbalanced Connections

Generally speaking, audio interfaces tend to have balanced input and outputs, because they are primarily intended for professional audio applications. However, this is not universally true; connection types vary among interfaces, and many devices even have different connections among input and output channels. For example, we have seen the following variations from balanced I/O on audio interfaces we have evaluated:

- Unbalanced outputs
- Auto-sensing ¼" phone jacks on the input and/or output that automatically switch to balanced when a TRS connector is inserted and to unbalanced when a TS connector is inserted.
- Connectors intended for software-selectable line level or instrument level inputs. Each input option has a different input impedance and gain.

- A ¼" TRS phone jack output connector which doubles as a stereo headphone jack or two unbalanced line level outputs. A special ¼" TRS phone plug to two unbalanced connectors is required to use the line level out mode.

Caution!

Given the variety of connector types and behaviors in use, be careful to understand each connector's characteristics, and always calibrate using the same connector type that will be used for testing.

Basic Calibration

A basic calibration of an audio interface can be performed in a few simple steps using a precision AC volt meter to measure the voltage output by the device. The procedure is:

1. Connect an output of the audio interface to the volt meter. For balanced connections, make sure that the volt meter is floating (both input leads are isolated from ground).
2. With the output Scaling Mode in APx set to digital, generate a signal at a specific level (e.g., 1.0 FS is convenient).
3. Note the measured voltage on the channel and repeat the process for each output channel. Then calculate the average of all channels (e.g. 3.370 Vrms).
4. Divide the voltage measured in step 3 by the digital level used in step 2 (e.g. $3.370 \text{ Vrms} / 1.0 \text{ FS} = 3.370 \text{ V/FS}$).
5. Change the output connector Scaling Mode to Analog and enter the number calculated in step 4 as the Scaling Factor. (Note: the scaling factor can be set in terms of RMS or peak level – make sure to apply the sensitivity in the unit matching the volt meter unit.) The ASIO output is now calibrated in volts.
6. Connect the ASIO device outputs to the ASIO device inputs, and set the input Scaling Mode to Digital.
7. Generate a known voltage (e.g., 1.0 Vrms) and record the measured level in FS on each input

¹ The reduction in voltage is approximately 50%, but the actual value depends on the impedances of the connected devices.

channel (e.g., 0.110 FS). Divide the measured level by the generator level (e.g., 0.110 FS / 1.0 Vrms = 0.110 FS/V).

8. Change the input connector Scaling Mode to Analog and enter the numbers calculated in step 7 as the Scaling Factor for each channel. (Again, make sure to use the same unit.) The ASIO input is now calibrated in volts.

Qualifying an Audio Interface

With low residual noise and distortion, and a comprehensive suite of audio measurements, an audio analyzer is the best tool to qualify an audio interface. In addition, AP audio analyzers come with certified calibration according to international quality standards, and even the least capable analyzer in the APx500 family has significantly better performance than most audio interfaces.

In the following sections, we show the results of qualifying a 2-channel audio interface intended for home recording studios or mobile recording. This device, which retails for less than \$200 USD, has two input channels, two output channels, 24-bit resolution and supports sample rates up to 192 kHz.

Qualifying the Audio Interface Input

Input Connections

Connect the audio analyzer's balanced outputs to the interface's input channels. Our sample audio interface has two XLR combo jacks. We're using ¼" TRS phone plugs to use the line input connection, because on this interface, using the XLR part of the combo jacks connects to the microphone preamplifier circuit.

Interface Adjustments

Knobs or sliders for setting Gain, Mix, Pan, etc.—whether physical controls on the front panel or software controls in a mixing application—should be set to the position in which tests will be conducted. In the case of physical knobs, the most repeatable settings are fully counter-clockwise or fully clockwise. For our sample audio interface, we set the gain knobs fully counter-clockwise (minimum gain).

Generator Test Levels

Choose an appropriate generator level for testing. For this example, most measurements were conducted at a level of +4 dBu (1.228 Vrms), a common reference level for professional audio recording. The other level used for

measurements such as signal-to-noise ratio is the device's specified maximum Line Input Level at Minimum Gain, +24 dBu (12.28 Vrms).

Gain (Input Scaling Factor) Measurement

With the input ASIO Scaling Mode set to Digital, measure the gain at +4 dBu using the Level and Gain measurement. For our sample audio interface, we measured:

Channel 1	64.07 mFS/Vrms
Channel 2	62.71 mFS/Vrms

These are the analog Scaling Factors used to calibrate this device's inputs. When the Scaling Mode is switched to Analog and these Scaling Factors are entered, with a generator level of 1.0 Vrms, the RMS Level measured on both channels matches the generator voltage to 3 decimal places, verifying the input calibration.

Once the calibration has been verified, the remaining measurements can be conducted either in Analog or Digital scaling mode, per user preference.

Input Frequency Response

Frequency response is a very important audio quality metric. Figure 4 shows the response of our sample audio interface at 192 kHz sample rate, as measured with the APx Frequency Response measurement, which uses the fast log-swept sine (chirp) technique. This device is well within the manufacturer's specified flatness of ± 0.2 dB from 15 Hz to 22 kHz. Note that the analyzer's high pass filter setting was set to DC-coupled for this measurement, to avoid conflating the results with the response of the analyzer's AC coupling filter.

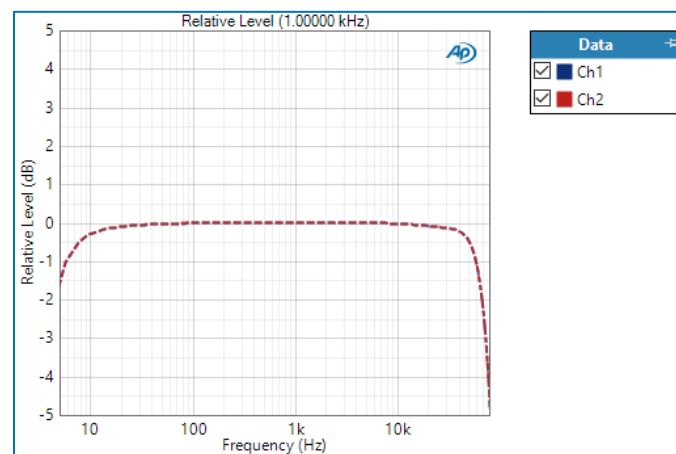


Figure 4. Relative Level response of the example audio interface from 5 Hz to 80 kHz, measured using the APx Frequency Response measurement.

Maximum Input Level

Our example audio interface is specified to have a maximum input level of +24 dBu at its minimum gain setting. However, a Stepped Level Sweep from -60 to +24 dBu (Figure 5) reveals that the distortion increases rapidly from about 0.001% at +19 dBu to over 6% at +24 dBu. Based on this result, we would hesitate using this interface for audio voltage level measurements greater than about +20 dBu (7.75 Vrms).

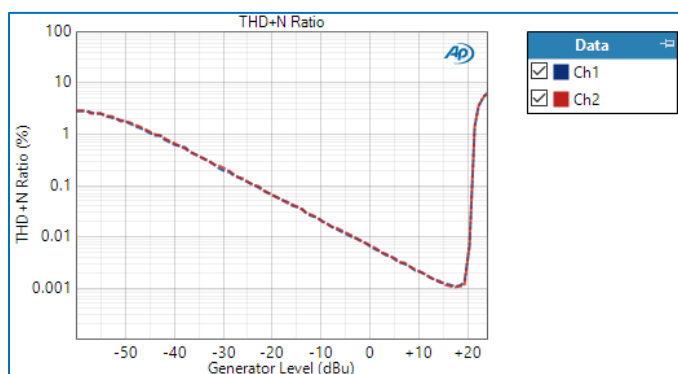


Figure 5. Input THD+N ratio versus generator level measured using a stepped level sweep from -60 to +24 dBu.

Input Noise, SNR, Crosstalk and Phase

These remaining audio quality metrics round out our audio interface input qualification. Note that the signal-to-noise ratio (SNR) measurement was conducted at the revised maximum input level of +20 dBu.

Noise (20 Hz to 20 kHz)	3.2 μ Vrms
SNR (+20 dBu at 1 kHz)	103.7 dB
Crosstalk (+4 dBu at 10 kHz)	-82.4 dB
Interchannel Phase (1 kHz)	0.01 deg

Having qualified the input system to this degree we now know exactly over what frequency range and input voltage range we can confidently measure input signals with this audio interface.

Qualifying the Audio Interface Output

Output Connections

Connect the audio interface outputs to the audio analyzer's balanced inputs. For our sample audio interface, we used ¼" TRS plug to XLR male balanced cables to connect at the interface's line out connectors.

Interface Adjustments

Here again, knobs or sliders for setting Volume, Mix, Pan, etc.—whether physical controls on the front panel or

software controls in a mixing application—should be set to the position in which tests will be conducted. In the case of physical knobs, the most repeatable settings are fully counter-clockwise or fully clockwise. For our sample audio interface, we set the Monitor and Mix knobs fully clockwise and an Output slider control in the device's software control panel to 0 dB (maximum output level).

Maximum Output Level and Linearity

On the output side, like any digital device, the maximum signal an audio interface can generate is 0 dBFS (or 1.0 FS). Since it's impossible for the signal to clip in this case (i.e., to exceed 0 dBFS) it's instructive to look at the output over the entire output range. Figure 6 shows the output THD+N ratio versus generator voltage as the output level is swept from -60 to 0 dBFS in 1 dB steps. Figure 7 shows the output linearity relative to a generator level of -20 dBFS for the same stepped level sweep. These results show that the THD+N is dominated by noise rather than harmonic distortion at all levels, and the output is quite linear over this wide output range. Hence, for this device, measurement can be conducted at any level up to the maximum level of 0 dBFS.

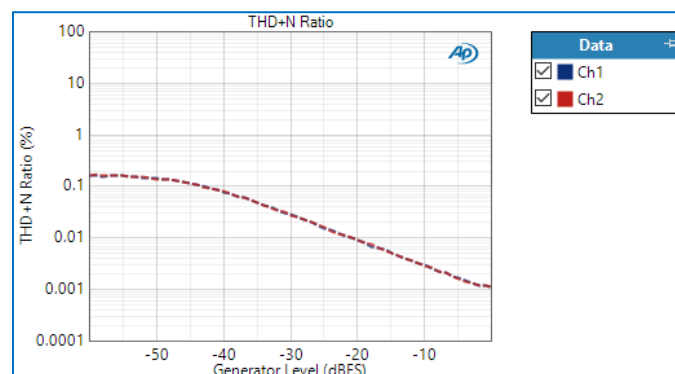


Figure 6. Output THD+N ratio versus generator level measured using a stepped level sweep from -60 to +0 dBFS.

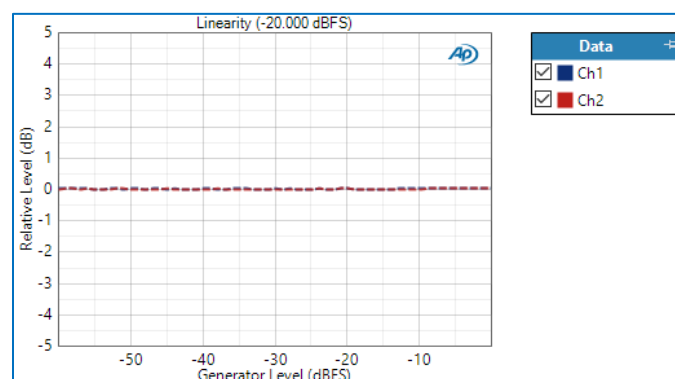


Figure 7. Output linearity relative to -20 dBFS for the stepped level sweep from -60 to 0 dBFS.

Our sample device has a specified maximum (nominal) output level of +10 dBu, which we confirmed with a level measurement of 10.332 dBu (2.545 Vrms) at 0 dBFS.

Gain (Output Scaling Factor) Measurement

With the output ASIO Scaling Mode set to Digital, measure the gain at 0 dBFS (1.0 FS) using the Level and Gain measurement. For our example audio interface, we measured:

Channel 1	2.555 Vrms/FS
Channel 2	2.535 Vrms/FS
Average	2.545 Vrms/FS

The average value is the analog Scaling Factor used to calibrate this device's outputs. When the Scaling Mode is switched to Analog and this Scaling Factor is entered, with a generator level of 1.0 Vrms, the RMS Level measured on both channels is within ± 0.004 Vrms (± 0.035 dB) of the generator voltage, verifying the output calibration.

Once the calibration has been verified, the remaining measurements can be conducted either in Analog or Digital scaling mode, per user preference.

Output Frequency Response

Figure 8 shows the response of the example audio interface at 192 kHz sample rate, as measured with the APx Stepped Frequency Response measurement. In this case, a stepped sweep was used instead of the log-swept sine measurement, because the latter is more sensitive to sample rate differences between the APx system clock and the audio interface. This device is well within the manufacturer's specified flatness of ± 0.2 dB from 15 Hz to 22 kHz.

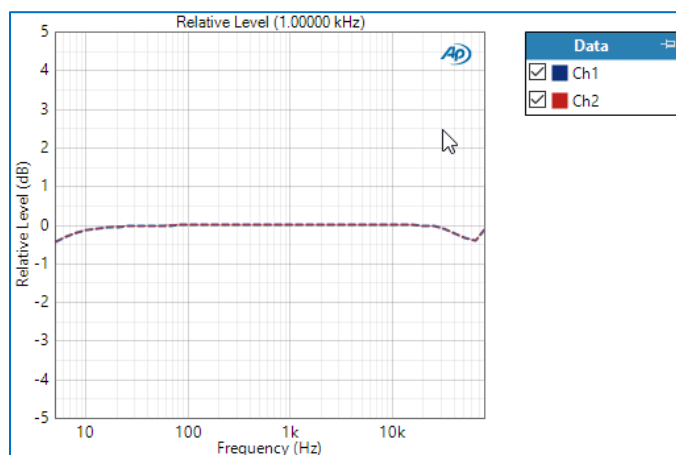


Figure 8. Relative Level response of the sample audio interface output from 5 Hz to 80 kHz, measured using the APx Stepped Frequency Response measurement.

Output Noise, SNR, Crosstalk and Phase

These remaining audio quality metrics round out our audio interface output qualification. Interestingly, this device had measured crosstalk of -118 dB for Channel 2 and -99 dB for Channel 1. However, both channels meet the specified crosstalk level of < -80 dB.

Noise (20 Hz to 20 kHz)	49.0 μ Vrms
SNR (0 dBFS at 1 kHz)	94.3 dB
Crosstalk 0 dBFS at 10 kHz)	-99.0 dB
Interchannel Phase (1 kHz)	0.008 deg

Having qualified the output system to this degree we now know exactly over what frequency range and output voltage range we can confidently generate signals with this audio interface.

Qualifying the Interface in Loopback

Finally, we qualify the audio interface in "loopback" (i.e., with its analog outputs connected to its analog inputs). In this mode we will test for glitches and dropouts, delay stability and delay reporting.

Glitches and Dropouts

Audio interfaces use a buffer of a prescribed length in samples to transmit audio data between the device and the PC. The buffer size may be selected from a list in the device's ASIO control panel. Buffer sizes vary from device to device. For example, the audio interface tested in this Technote has buffer sizes from 32 to 8192 samples.

The buffer size affects latency (the time between the audio signal's occurrence and when it is available for processing in the PC), and CPU usage. When an audio interface is used for recording or monitoring live music, low latency (a small buffer size) is desirable. However, if the buffer size is too small, audio samples can be lost as the PC's CPU tries to keep up with transmitting the audio on top of other tasks. This can result in glitches and dropouts, which may be audible as pops or clicks. Regardless of whether these glitches are audible, they are definitely undesirable when an audio interface is being used for audio test. Low latency is not important for most audio tests, so the buffer length should be increased to the point where glitches and dropouts do not occur.

The APx500 software has a measurement named Measurement Recorder, which is very useful for detecting audio glitches and dropouts. This measurement generates a signal and records audio metrics such as RMS Level,

Frequency, THD+N, etc., versus time with a reading rate of up to 20 readings per second in Sequence Mode (up to 250 per second in Bench Mode). If a sine signal is generated, the THD+N Ratio result is extremely sensitive to glitches and dropouts. Even a loss of a single sample at a sample rate of 192 kHz will cause a large spike in the THD+N vs time graph. For example, Figure 9 shows the THD+N versus time graphs recorded for 60 seconds for our sample audio interface in loopback mode with ASIO buffer sizes of 512 and 2048 samples. As shown, with a buffer size of 512, many glitches occurred within the 60-second recording, whereas with a buffer size of 2048 samples, there were none. Clearly the 512-sample buffer size is too small for audio tests with this device.

The time period over which glitches should be checked for depends on the application. For example, if you are conducting short term audio tests that take less than 30 seconds, you would want to ensure the device is glitch free for at least that length of time. However, if you're testing a device for glitches over several hours or days, you need to ensure that the audio interface itself is glitch free for that long. Measurement Recorder in APx can be run for up to 7 days, if required.

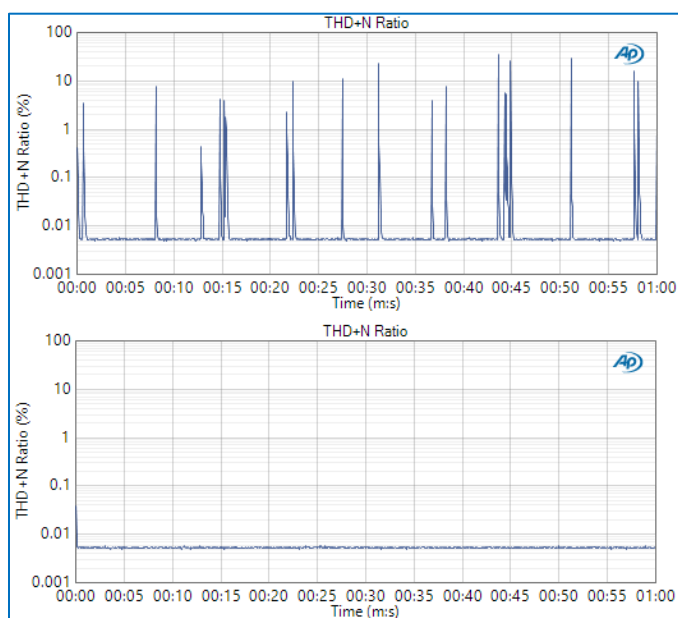


Figure 9. THD+N Ratio vs Time measured with an audio interface in loopback, with ASIO buffer lengths of 512 samples (upper graph) and 2048 samples (lower graph).

Delay Reporting and Delay Stability

Once the audio transmission has been verified to be glitch free, we can test for delay reporting and delay stability.

ASIO device drivers can be queried to find their input and output latency in samples. When the Latency Auto checkboxes in the ASIO input or output configuration panel in APx are checked, the software uses this feature to automatically compensate for the device's reported delays when generating or acquiring signals. When the Auto checkboxes are not checked, the user can enter the input or output latency in samples.

If the device's delay reporting is correct, when the audio interface is in loopback mode, the delay measured through the device should be zero seconds. The measured delay should also be stable, such that when it is measured several times, the measurement results are equal.

The APx500 software has a measurement called DUT Delay which is used to measure the delay through a device under test. It works by generating a pseudo-random signal through the device and using cross-correlation of the DUT output signal with the original signal to find the delay.

For our sample audio interface, with the device in loopback and the input and output Latency set to Auto, the DUT Delay measurement consistently measures a delay of $-41.67 \mu\text{s}$. The consistent result means the delay is stable, which is the desired result. However, the fact that the delay is negative indicates that the device's delay reporting is wrong.

A delay of $41.67 \mu\text{s}$ at 192 kHz sample rate corresponds to 8 samples. Either the reported input or output latency (or a combination of the two) is wrong by 8 samples. This can be rectified by unchecking the Latency Auto checkbox of either the input or output and adjusting the reported Latency up or down by 8 samples. For our sample audio interface, we unchecked the input Latency Auto checkbox and reduced the Latency from the reported value of 2912 samples to 2904. Once this was done, the DUT Delay consistently returned a result of 0.0 seconds.

Using the Audio Interface for Audio Test

Having completed a comprehensive set of measurements to qualify the audio interface's outputs and inputs, and to ensure glitch-free operation and delay compensation, we are ready to put the device to work conducting audio tests.

When using an audio interface for audio test, it's extremely important to ensure that:

- All physical and software controls (knobs, sliders, numeric controls, etc.) for setting Gain, Pan, Mix Volume, etc. must be in exactly the same position in which the

audio interface was qualified and calibrated. Some users go so far as to break the gain or volume knobs off the front panel of an audio interface or to otherwise lock them in place, so gain settings can't inadvertently be changed, resulting in incorrect measurements. Also, some devices have the ability to lock their software control panels with a password. This feature should be used if available.

- Connection types and physical connectors used for audio test should be exactly the same as those used to qualify and calibrate the audio interface.

Interface vs Hardware Analyzer

Pros and Cons of an Audio Interface

In addition to having a much lower price point than a physical audio analyzer, there are some other advantages to using an audio interface for audio test, depending on the application, including:

- Many audio interfaces have one or more microphone preamplifier inputs which typically provide phantom power for condenser microphones. Users who need to make acoustic measurements can leverage this by using phantom powered measurement microphones such as Audio Precision's 376M03 (Figure 10).
- Most audio interfaces have a stereo headphone output jack with sufficient power to drive typical headphones. This can be leveraged for users who need to conduct audio tests of headphones and earphones.
- Audio interfaces typically have a smaller form factor than a hardware analyzer, making them more convenient for applications needing portability. Some are even powered via USB.



Figure 10. Audio Precision 376M03 phantom powered measurement microphone.

The biggest downside to using an audio interface for audio test is how easy it is to inadvertently destroy the device's calibration. They typically have several physical or software knobs, sliders or numeric controls, which if changed will put the device out of calibration resulting in erroneous

measurement results. Additionally, simply using a different input or output connection type than the connections used for calibration can also result in incorrect measurement results.

Pros and Cons of a Hardware Audio Analyzer

As the manufacturer of the world's leading audio measurement hardware, we at AP are no doubt biased in this regard, and unapologetically so. There are a multitude of reasons why a physical audio analyzer is better than an audio interface for audio test, including:

- Instruments designed for test and measurement with factory-certified calibration to international standards.
- Robust I/O circuitry with over voltage protection.
- A 3-year warranty on audio analyzer hardware.
- Measurement-focused, and experienced, technical support.
- Higher output and input voltages than an audio interface.
- Ability to generate and measure DC signals.
- Multiple input and output ranges with auto-ranging to optimize residual THD and noise performance.
- Ability to add optional digital I/O modules such as Bluetooth, HDMI, PDM and digital serial (DSIO).
- Specialized audio measurements such as Common Mode Rejection (CMR), etc.
- Jitter generation and measurement.
- A convenient internal loopback path is available on many analyzer models.
- GPIO ports for controlling and sensing external switches and relays, etc.
- Self-test procedures which enable users to verify instrument performance in the field.
- Express lack of physical knobs or sliders that facilitate measuring incorrect data.

In terms of disadvantages, physical audio analyzers have a larger form factor than an audio interface, making them less portable. And, as noted in the opening paragraph of this technote, tend to be more expensive than audio interfaces.

Conclusion

We've covered the measurements needed to calibrate and qualify an audio interface for audio testing, and we've identified some of the pitfalls involved. Hopefully, this will be helpful to users who want to use an audio interface other than one of the three models our engineering team has verified as compatible with APx500 Flex.

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