ADC Performance Testing
Report on Project Development During 2015

Expert Consultant Report for the
Audio-Visual Working Group,
Federal Agencies Digitization Guidelines Initiative

Topics:

- **Refinement of the Comprehensive High Metrics Guideline and Associated Test Method**
- **Initial Development of a Partial Minimum Metrics Guideline and a Low Cost Test Method**

avps

253 36th Street, Suite C302
Brooklyn, New York 11232
917.475.9630

FADGI dissemination draft prepared
February 15, 2016

Table of Contents

I. Introduction

Background

The Federal Agencies Digitization Guidelines Initiative (FADGI) Working Group has been exploring the performance testing of audio digitization systems since 2012. This topic includes two main elements: (a) the performance of analog-to-digital converters (ADCs) and (b) the problem of interstitial errors, i.e., accidental loss or transformations of audio samples within the digitizing system before the data stream is written to file. Both elements are of high interest to FADGI member agencies and also respond to recommendation 2.4 of the *National Recording Preservation Plan*, "Preservation Workflows for Audio Materials."[1]

This report provides information on the metrics relevant to the measurement of the performance of analog-to-digital converters (ADCs): (a) what to measure, (b) how to measure, and (c) what are the pass-fail points? The main author is the Working Group's expert consultant Chris Lacinak of Audiovisual Preservation Solutions, supported by the audio engineer Phillip Sztenderowicz,[2] with additional contributions from Carl Fleischhauer, the coordinator of the FADGI Audio-Visual Working Group.

The report builds on earlier FADGI guidelines and reports,[3] including the following:
February 2012. *Analog-to-Digital Converter Performance Specification and Testing*. The initial explanatory report and proposed guideline.[4]
August 2012. Two documents:

- *Analog-to-Digital Converter Performance Specification and Testing*. The guideline as approved on that date.[5]
- *Audio Analog-to-Digital Converter Performance Specification and Test Method: Introduction*. An introductory document, updated from an earlier version.[6]

The report also builds on the important *Guidelines on the Production and Preservation of Digital Audio Objects* (TC-04, Second Edition, 2009; International Association of Sound and Audiovisual Archives), and several international standards, including *AES standard method for digital audio engineering — Measurement of digital audio equipment* (AES17-1998, r2009; Audio Engineering Society)[7] and IEC-61606-3: *Audio and audiovisual equipment - Digital audio parts - Basic measurement methods of audio characteristics - Part 3: Professional use; Edition 1* (International Electrotechnical Commission).[8]

---

[1] http://www.loc.gov/programs/static/national-recording-preservation-plan/publications-and-reports/documents/NRPPLANCLIRpdfpub156.pdf
[2] Sztenderowicz participated in this project under the auspices of Audiovisual Preservation Solutions; he also works as a technical engineer at Sterling Sound in New York.
[3] Links to all relevant FADGI documents are provided here: http://www.digitizationguidelines.gov/guidelines/digitize-audioperf.html
[4] http://www.digitizationguidelines.gov/audio-visual/documents/ADC_Perf_Test_2012-02-24.pdf
[5] http://www.digitizationguidelines.gov/audio-visual/documents/ADC_performGuide_20120820.pdf
[6] http://www.digitizationguidelines.gov/audio-visual/documents/ADC_performIntro_20120820.pdf
[7] http://www.aes.org/publications/standards/search.cfm?docID=21
[8] https://webstore.iec.ch/publication/5666

As the work has moved forward, various individuals have asked, "Why carry out ADC performance tests in the first place?" They sometimes add the argument that manufacturers provide good specifications for their equipment as a part of their marketing. In response, the expert consultants examined the specifications from several ADC manufacturers, and saw that these often fail to provide complete statements of what has been measured or about the test methods employed.

Another question heard during the development period asked, "Is it necessary to test solid-state devices like ADCs an ongoing basis?" The FADGI response is "yes." In part this is a matter of common sense, but it is also the case that the expert consultant had informal conversations with other experts, including members of the Audio Engineering Society committee on Digital Audio Measurement Techniques. Their comments were very consistent. Every expert favored routine testing, arguing that ADCs are no different than any other type of equipment and stating that these devices can fail in nuanced and subtle ways.

Conceptual framework

The work carried out in 2015 confirmed some of the emergent findings of FADGI's prior activities on ADC performance testing. These findings represent the intersection of two factors. The first factor is the Working Group's interest in allowing for *levels* of performance. The 2012 guideline pertains to ADC performance at the highest level. From the start, however, the Working Group also conceived of guidelines for lower levels of performance as well, potentially labeled moderate and minimum. A lower level guideline might be selected by organizations that command modest resources but wish to proceed with a digitization project, e.g., a federal agency with a historical collection of recordings of lecture-like presentations by staff originally recorded on audiocassettes. The agency may determine that "very good" digitized copies produced with a moderate performance system will meet every conceivable future need. Or an archive may have certain classes of material, e.g., radio station logging tapes, for which "acceptable" digitized copies produced with a minimum performance system will be sufficient.

The second factor has to do with the cost of testing equipment. The 2012 guideline contains 12 metrics, several with very exacting measurements. The ADC testing thus far has demonstrated that the evaluation of all 12 metrics at the desired levels of precision requires an audio analytic device from a class that costs upwards of $20,000. Several large federal agencies possess such devices (two participated in the 2015 field test) but many others do not, and the cost is prohibitive for most of them.

The combination of the preceding factors led the consultants and the Working Group to explore two types of systems:

- System capable of comprehensive ADC testing against all 12 metrics at a high level of performance, regardless of cost.
- System capable of ADC testing against a subset of the metrics at a minimum level of performance, at low cost (less than $1,500). Such a system developed during the 2015 round of work proved to be capable of testing against 7 of the metrics.

Regarding the second type of system, where lower-cost test and measurement components rule out testing all 12 metrics at a high level of performance, the Working Group felt that some testing was better than none. The discipline of performance-testing an ADC leads people to pay close attention to a variety of factors, including ones not being directly tested, thus increasing the likelihood that the technical aspects of a digitization facility will be properly set up. The Working Group wishes to encourage this outcome.

As the 2015 system testing proceeded, the expert consultants found themselves considering a third system type that should also be developed:

- System capable of ADC testing against a subset of the metrics at a high-to-moderate performance level, at low-to-moderate cost.

Such a system would probably cost a bit more than the low-cost system tested in 2015. However, the advantage of such a system is the ability to test an ADC for higher levels of performance, presumably against more than the 7 metrics identified for the minimum performance system. And if such a system carried a low enough cost, its existence would eliminate the need for the low-cost, minimum performance test system. In other words, a moderate level test system could test ADCs at the moderate and minimum performance levels. For the 2015 round of work, however, scarce resources meant that the development of a moderate performance level system was deferred for a future year.

Considered together, the three system types provide a conceptual framework that guided the expert consultants as they carried out the 2015 round of work, and as the Working Group laid plans for the 2016 effort. However, the 2015 activities highlighted an unanticipated factor that has influenced the framework.

This unanticipated factor has to do with the intricacies of measuring audio performance, and the ways in which the specific measurement tools--e.g., high cost, low cost--execute the measurements, also known as test methods. The fact is that a low cost system may provide a reasonable assessment of, say, Intermodulation Distortion (IMD), but (as described later in this report) it does not do so in precisely the same manner as the high cost measurement system. Thus, although both systems offer a measurement of IMD, and although the low cost system's ability to measure that performance "tops out" at a lower quality level than the high cost system, it is not strictly the case that there is a simple "lower performance number" relationship when compared to the reading from the high cost system. It is also the case that the low cost system itself has limited capabilities. The ADC being tested might perform better than what the low cost system reports, due to the testing system's limitations. However, a low cost test system will identify more extreme types of failures of an ADC, even if it is a high performance unit.

The outcome of the preceding analysis is represented in the table that follows. This conceptual framework uses terms to name the test systems that reflect both of the elements in play: *measurement system cost* and *ADC performance*.

Conceptual Framework

Test System	Number of metrics	Performance guideline status
Capable of confirming high quality ADC performance, high test-system cost Included in 2015 activities	12	Guideline approved in 2012; some revisions proposed in section VI of this report
Capable of confirming moderate quality ADC performance, low test-system cost Not included in 2015 activities	8-9 (to be determined)	To be determined during 2016 activities
Capable of confirming minimum quality ADC performance, low test-system cost Included in 2015 activities	7 (proposed in this report)	New draft guideline proposed in section VI

The following bullets provide some particulars for the 2015 activities in terms of the preceding conceptual frame, and using slightly rearranged terminology in the test-system naming. These particulars are further elaborated in sections III, IV, and V.

- **ADC Test System (High Metrics, High Cost).** The benchmark for this is the system employed in the 2015 project, consisting of the Audio Precision SYS 2722 together with other elements. The SYS 2722 is typically sold at prices ranging from $20,000 - $25,000. The results of the 2015 testing are in this report. This system is capable of performing a comprehensive test, although in the 2015 proof-of-concept tests, the actions required a fair amount of manual intervention and multi-step data processing to generate an assessment report.
- **ADC Test System (Moderate Metrics, Low Cost).** Not tested in 2015. FADGI and Library specialists speculate that certain types of commercial signal generators should support this level of testing with success. Such devices for professional use carry costs on the order of $1,500-2,500, and supporting software packages cost in the range of $100-200. Thus a partial system would be significantly less expensive to develop than the comprehensive system.
- **ADC Test System (Minimum Metrics, Low Cost).** Initial testing was carried out in 2015, with the use of the NTi MR-PRO Minirator Audio Signal Generator (about $600) and the SpectraFOO and ARTA software packages (on the order of $100-200). The results of the 2015 testing are in this report. Although more constrained than the higher levels, this test will be valuable and, as noted above, the simple act of executing it may reveal unrelated operational problems to facility staff.

Arrangement of this Report

This report discusses the findings that resulted from development and testing during 2015. The sections of the document are arranged more or less in chronological order, although the activities are described as if they had proceeded in a linear fashion. In fact, several of the steps overlapped each other or occurred in parallel, but these calendar facts are not material to the technical information presented here. Summary conclusions are presented in section VII.

II. Setup for the Comprehensive High Metrics System

Purpose for the setup

The expert consultants (Lacinak and Sztenderowicz) assembled a proof-of-concept setup capable of testing ADCs against the 2012 FADGI high-level performance guideline. The SYS-2722 device supports the use of scripts that can be used to direct the system to perform a specific set of tests, and the setup included the creation of a series of scripts that guided the system through a set of actions that are appropriate for the testing of ADC devices against the FADGI guideline. This proof-of-concept system, however, was not required to have fully realized operational software nor a user-friendly interface.

Once assembled, the proof-of-concept system was used to perform an initial series of tests at the consultant's facility on a high-quality ADC. As will be reported below, a number of challenges and difficulties were encountered although, in the end, the assembled system performed in a manner sufficient to move through the phases of the activity.

Basic System Setup

After receiving the Audio Precision (AP) 2722 from the Library of Congress the unit was setup in a test suite using a stabilized power system that was monitored using a volt meter and frequency counter. For control and reporting the 2722 was connected via the AP serial-to-USB converter to an HP laptop running Windows 7 with the corresponding AP software installed. The 2722 was also connected to a high-quality, professional ADC via balanced XLR and AES cables. The ADC in these setup tests "stood in for" the ADCs that would be field tested later in federal agencies. Figure 1 illustrates the setup.

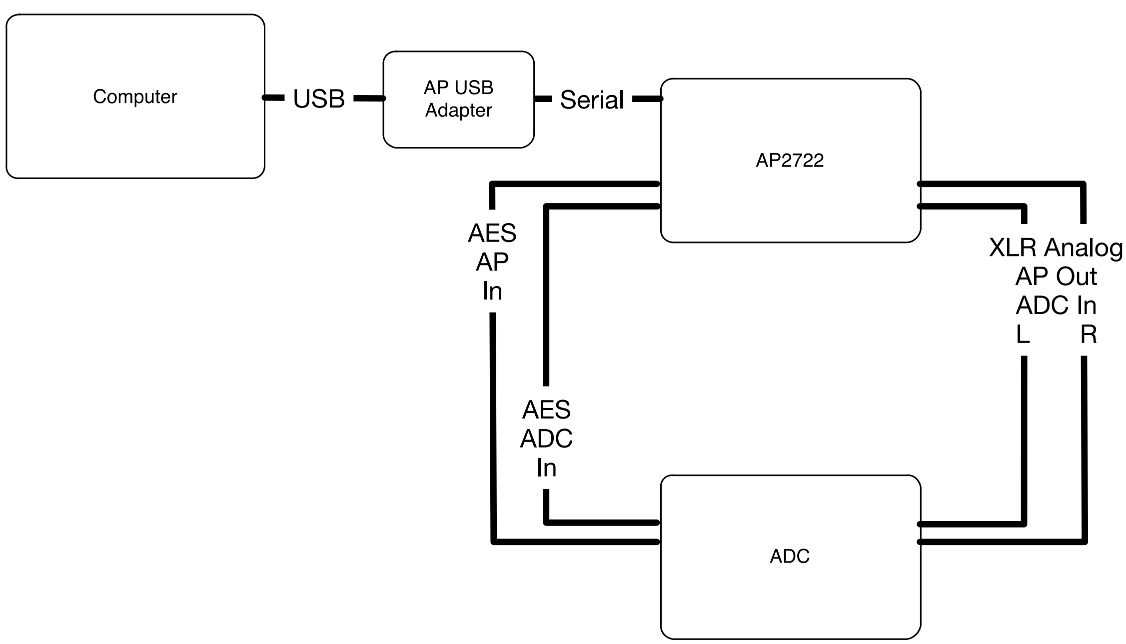


Figure 1: Basic Diagram of Test Setup

Figure 2: Image of the AP2722 and computer

Figure 3: Phillip Sztenderowicz with the AP2722, computer and the setup-test ADC

A spreadsheet was created for preliminary tracking and documentation of tests. This included the date, time, engineer name, device name, device serial number, ambient temperature of the room, and the mains voltage, test name, test result. Before performing testing, the ADC and AP were powered on and allowed to warm for at least 30 minutes and the temperature of the device was measured and documented prior to testing.

Following basic familiarization the consultants began running pre-programmed tests that came as stock tests within the AP software in order to get a better handle on the operation and reporting of the 2722. These include tests such as frequency response and total harmonic distortion + noise (THD+N).

Challenges

Once acquainted with the basic operations of the 2722 the consultants began to explore implementing the guideline using the device. This raised many questions. Generally speaking, it began to reveal portions of the test method which were lacking clarity, inviting multiple interpretations on how to perform a given test. This manifest in two ways. The first was in reading the guideline and realizing that the precise settings to be used on the 2722 were not evident. The second was differing results compared to the 2012 study that was conducted in support of the initial drafting of the guideline. The consultants also found that reporting using the 2722 was more difficult than originally anticipated, requiring a combination of external scripting and advanced internal scripting using AP's language referred to as AP Basic.

Specifically, the consultants confronted the following issues:

- **Automating Pass/Fail Reporting**
  It was challenging to figure out automating the reporting of pass/fail results for the range of tests. This required using what's referred to as tabular sweeps, capturing tabular data and then comparing against established thresholds at a particular point.

- **Automating Reporting in Reference to the Guideline**
  The consultants found that automating pass/fail reporting was a distinct activity from reporting in reference to the guideline performance metrics. For instance, the CMRR test method states the following:

  > *The resulting RMS value, measured in dBFS, is increased by 20 dB and reported as a dB (not dBFS) value.*

  Additionally, the performance metric states the following:

Frequency	***Limit***
60 Hz	*70 dB*
1 kHz	*70 dB*
20 kHz	*50 dB*

However, the results within the AP device are reported as the original value prior to the transformation spelled out in the test method, which is necessary to arrive at the value referenced in the performance metric. Therefore, the pass/fail reporting in the AP is based on the original non-transformed value, but the reporting for the sake of the performance metric is the transformed value. This meant that the consultants needed to perform additional steps in order to report according to the guideline, even after the consultants had pass/fail reporting in place, and this was true for many tests.

- **HF IMD 2kHz Spread**
  The guideline states that the frequency spread should be 2kHz but the 2722 IMD testing does not allow this. The consultants had to devise an alternative mechanism for measurement.

- **LF IMD 41Hz**
  The analog signal generator would not allow us to generate a 41Hz tone. The consultants were concerned that the digital generator would not be of sufficient quality but testing revealed that it was high enough quality to use.

- **CMRR**
  The test method indicated that 600 Ohms of resistance should be used, which was out of alignment with the most recent IEC and AES revisions of CMRR testing. Using 600 Ohms of resistance also produced different results than the documented results from the first round of testing in 2012. When using 10 Ohms instead of 600 Ohms the consultants found the test results to align with the 2012 results. In reviewing testing notes from 2012 the consultants were able to identify the decision to use 10 Ohms instead of 600 Ohms, confirming our findings.

  Additionally, this test required building a specialized cable, inserting 10 Ohms of resistance on each leg of a balanced analog cable independently between the AP2722 and the ADC. This cable can be seen in the image below. There are four buttons in the image. Two of them will insert 600 Ohms of resistance and two of them will insert 10 Ohms of resistance into each of the legs on a balanced cable. Because there was a period of time when the consultants were not sure which one was correct they built a cable with both options. Future cables would not require the 600 Ohms of resistance, but such a cable is necessary for performing this test.

Figure 4: CMRR cable built by Phillip Sztenderowicz

- **Cross-Talk Band Pass Filter**
  Cross talk tests typically use a band pass filter on the channel being measured. The consultants noticed that the test method did not explicitly state the use of a band pass filter and their initial results were not accurate.

- **Amplitude Linearity Standard Deviation**
  Our guideline settled on using standard deviation for the main metric. This differs from other amplitude linearity performance metrics and the amplitude linearity test built into the 2722 which looks at peak deviation. In putting together the original guideline the consultants felt that standard deviation gave a fairer and more accurate point of comparison. However, the 2722 does not have a way to report on standard deviation, requiring the tabular data from this test to be exported and calculated externally.

- **Spurious Aharmonic Signals**
  In reading the test method the consultants realized that it was unclear how to perform this measurement. They used a 32k FFT with a Rife Vincent 5 window, power averaging 8 FFT buffers. After doing this the consultants felt that filtering out the stimulus harmonics completely prior to performing the measurement was the most accurate way of doing this but it was not mentioned in the test method. They were also unclear about the range within which they should perform the measurement. The consultants questioned whether measurement should be performed below 1kHz due to the low chance of there being spurious signals beneath 1kHz, the likelihood that the issues found below 1kHz would be power supply noise, and the probability that a clocking issue that created a spurious signal below 1kHz would be mirrored above 1kHz as well. And finally, they felt that it should be made explicit in the test method that the measurement and reporting should be the absolute value of the largest spurious aharmonic peak.

- **Sync Input Jitter Susceptibility**
  Early on there was some confusion that arose because the guideline references AES-17, which uses a THD+N methodology to measure the effect of jittering the clock input. However, the results being reported in the first round of testing in 2012 were better than the un-jittered clock THD+N performance for the converter. This is illogical. Furthermore, the guideline, after referencing the AES-17 methodology, states:

  > *The output spectrum is measured at each step and the results overlaid. …. Results are expressed as dBFS for each octave step.*

  Ultimately it was unclear how the measurement should take place. Upon review of notes from 2012 the consultants were able to surmise that the output should be measured using an FFT. For a given stimulus (12KHz and 997Hz at the ADC inputs); the ADC’s reference clock is then jittered in octave steps over a range of frequencies from 63Hz up to one step below the stimulus frequency. The peak values of the resultant artifacts that appear at the sum and difference of the jitter frequencies relative to the fundamental stimulus (12KHz and 997KHz) are then measured: i.e. the peak values that appear at 11,937Hz and 12,063Hz for a 12KHz input with the clock jitter frequency being 63Hz; or 497Hz and 1497Hz for a stimulus of 997Hz at the input with the clock being jittered at 500Hz.

The difficulty encountered raised some insights that were outside the scope of the setup activity itself. Phillip Sztenderowicz, the co-investigator on this project is a veteran bench technician that has worked at Prism Sound, a manufacturer of ADCs and test and measurement equipment, and he has performed maintenance at world-class audio facilities for over two decades. The co-investigator on the initial effort was Richard Cabot, another test and measurement veteran who was a founder of Audio Precision, part of the original group that drafted AES-17, and is widely known in the test and measurement community.

One of the original aims of this effort was to create a *simple* tool that could be used by non-experts. The exercise of having one expert interpret and apply the specification of another, even when based on standards, was enlightening in regard to the remaining obstacles to achieving use by non-experts. The consultants found many instances in the test method in need of clarification. Some examples of this are above, but will result in specific recommendations to modification of guideline language in a later report. They also discovered the fact that there is an important distinction to make between the *interpretation* of the standard and the *application* of the standard with a given test and measurement device. Once the language of the guideline is understood, they found that using a device as sophisticated and complex as the AP2722 presents its own challenges. They also discovered that there were multiple possibilities for applying some test methods, often generating differing results. This is particularly problematic because it is not about understanding language in the guideline, but rather having to do with the specificities of a given device. While a good guideline is clear and explicit in its language, it cannot be so specific as to speak to a single device. Naturally, there will be variances between manufacturers and even within manufacturers across different device models. The inherent implication of this is that the test method and reporting is applied differently with different devices. Additionally, the precision and accuracy required at this level of testing (beyond parts per million precision) produces very low margin for error. This paints a picture in which the complexity of the standard and device create a high probability of imprecision, while the stringency of the testing requires extreme precision. This realization drives home the need for simplicity, even at expert levels, but especially at non-expert levels, in order to fulfill the original vision of a simple tool that can be broadly adopted and used. It enforces the need for a “sole purpose” test setup that is purpose-built to perform the test methods and reporting according to the guideline. In the immediate term, and at minimum it bolsters the value of the work being done in this phase to clarify the guideline, create templates for the AP2722, and document the procedure.

Because of the challenges discussed above, the time required to perform the initial setup and testing of the AP2722 was well in excess of the original estimate. It was also not a discrete activity. In other words, the initial setup and testing process turned out to provide valuable insights into the overarching goals and objectives. As part of this process the consultants began to get glimpses into areas of the guideline that were unclear and had to begin working out the procedure much earlier than they originally imagined. In this way, the setup process turned out to be both more difficult and more integral to the larger effort than initially anticipated.

Advanced System Setup

Once the tests were fleshed out and the consultants were certain that the interpretation and application were correct, they embarked upon creation of test templates and scripts for the AP2722. The former involves establishing the parameters and setup of a given test and saving it as a test template. A template includes the stimulus type, frequency, level, sequence, etc. It also includes the method of measurement, such as FFT or level meters, pass/fail thresholds, the reporting and the layout of the windows. When you load the test template everything is setup and ready to run the test, but it still requires the user to run the test, capture the results and report. To automate this, AP scripts are necessary.

Following the creation of a test template for each test in the guideline the consultants drafted scripts to enable the automated running of each test. The script includes the automated loading of the template, calibrating the level, running the test, and reporting the results. Some tests are able to be fully automated but others can only be semi-automated due to physical setup change requirements. An example of this is the CMRR test in which the aforementioned specialized cable must be inserted and then each leg must be tested before identifying the leg to which resistance should be applied while running the test.

As mentioned earlier there is a distinction between running the test to get a pass/fail report and generating reporting according to the guideline's performance metrics. In some cases the consultants were able to use AP scripting to perform the calculations and report according to the guideline metrics within the AP environment. In other cases they had to use AP Basic scripting language to export the data to Excel and then use Microsoft Basic to perform the necessary calculations and reporting. This was unexpected and took a significant amount of time to tackle for some tests. In particular, calculating standard deviation for amplitude linearity proved to be time consuming.

There was a third level of automation that the consultants originally wanted to have in place prior to testing more broadly. This was to create an overarching script that would run each of the individual test scripts in a sequence from beginning to end with the push of a single button. They started down this path but ran out of time to adequately program and debug the script. Using the test-level scripts the entire test suite requires less than 10 minutes and they felt that this was more than sufficient for the purpose of efficiently performing the tests within the guideline at this phase.

Test Result Documentation

The AP2722 and similar devices are largely used for real-time testing and reporting: testing a device, identifying whether the device is working or not and responding accordingly in the moment. They are not setup by default to store or export the resulting data for longer-term documentation and evaluation. The longer term keeping of the results was seen as an important component of this work, causing us to dig deeper into the reporting capabilities and options for maintaining this information. The consultants identified three ways to do this. The first is to use the preference within the AP device to log all data. This places the text-based results in a simple text-based log file which can be parsed and evaluated at a later date. However, in order to save the data for a single device it requires the user to clear the log data before testing begins and to save the log file after the testing of a device ends. As mentioned earlier, part of our testing utilizes external applications such as Excel to perform calculations that report in reference to the guideline. This means that the Excel files generated during the testing for a given device also need to be saved to the directory where the log files are saved. Lastly, the graph data generated in the AP software is a valuable data point and reference. Keeping these requires exporting the images of the graphs to the same directory as the log file and Excel documents. The collection of these three items make up the total test result package. Ideally, an overarching macro could perform all of these actions, parse the appropriate data from each of the generated files, and create a master report exhibiting pass/fail status of each test along with the more detailed results and associated graphs. However, there was not enough time in this phase to accomplish this vision.

Test Results with the Stand-in ADC

Testing of the high-quality ADC selected for initial setup purposes using the guideline generated the following results:

Frequency Response
96k Hz Sample Rate

Frequency	Limit	ADC	
	dBFS	Left dBFS	Right dBFS
20 – 20k Hz	+/- 0.1 dBFS	-0.09 dBFS	-0.09 dBFS
20k - 40k Hz	+/- 0.5 dBFS	-0.053 dBFS	-0.051 dBFS

THD+N

Freq	Level	Limit	ADC	
Hz	dBFS	dBFS	Left	Right
41	-1	-100	-104.4	-103.8
997	-1	-100	-104.8	-103.2
6597	-1	-100	-104.5	-103.5
997	-10	-100	-102.4	-102.3
997	-20	-90	-92.5	-92.4
997	-60	-50	-52.7	-52.6

Dynamic Range

	Limit	ADC	
		Left	Right
Unweighted	-110 dB	-112.7	-112.4
A weighted	-112 dB	-115.2	-115.1

Crosstalk

Frequency	Limit	ADC	
		Left	Right
20 Hz	-110 dB	-121.8	-118.7
997 Hz	-110 dB	-135.4	-134.4
20 k Hz	-105 dB	-146.1	-146.1

CMRR

Frequency	Limit	ADC	
		Left	Right
60 Hz	70 dB	88.3	87.6
997 Hz	70 dB	85.3	85.1
20 k Hz	50 dB	61.5	62.1

LF IMD

	Limit	ADC	
		Left	Right
LF sum	-100 dB	-94.8	-93.6

HF IMD

	Limit	ADC	
		Left	Right
HF sum	-105 dB	-114.0	-111.7

Amplitude Linearity

	Limit	ADC	
		Left	Right
Standard Deviation	0.05 dB	0.047	0.019

Spurious Aharmonic Signals

Frequency	Limit	ADC	
		Left	Right
>50 Hz	-100	-129.0	-125.6

Alias Rejection

SR	Limit	ADC
96 k Hz	-80	-99.39

Jitter Susceptibility

12 k Hz

Frequency	Limit	ADC
8 kHz	-130 dB	-138.3
4 kHz	-120 dB	-138.1
2 kHz	-120 dB	-137.4
1 kHz	-120 dB	-131.2
500 Hz	-100 dB	-109.1
250 Hz	-90 dB	-92.5
125 Hz	-70 dB	-72.4
63 Hz	-60 dB	-66.2

997 Hz

Frequency	Limit	ADC
500 Hz	-110 dB	-133.1
250 Hz	-100 dB	-112.9
125 Hz	-90 dB	-94.3
63 Hz	-80 dB	-87.8

Jitter Transfer Gain

Limit	ADC
<20ns p-p	1.63ns

Operator Skill Level Requirement

The execution of an ADC test using the comprehensive high metrics system requires the supervision of someone with a strong knowledge of audio engineering and digital technology. A sense of this requirement is provided in section V, in the description of the field test, which includes a step-by-step script to guide the operator carrying out the test. To a degree, the path through this script could be made easier by the provision of software support, and the development of such software is planned for 2016. However, the intricacies of the test and the vagaries of operating the analyzer and ADC devices (each with quirks, as described in section V) mean that it is wise for this testing to be setup, confirmed to be functioning correctly, and generally supervised by an expert even if routinely performed by a non-expert.

III. Setup for the Partial Minimum Metrics System

Purpose for the setup

The expert consultants assembled a proof-of-concept setup capable of testing ADCs using the following low-cost devices and software. The setup employed an NTI Minirator MR-Pro as a signal generator and, for the analyzer, the expert consultants used ARTA software, which offered increased capabilities when compared to Spectrafoo and other similar software. For example, in this price class, ARTA was uniquely able to perform the high frequency and low frequency intermodulation tests.

Once assembled, the proof-of-concept system was used to perform an initial series of tests at the consultant's facility on selected ADCs. This preliminary testing led to the development of an overall metric approach, which was then further tested in the field during the visits to federal agencies described in the section devoted to that phase of the activity.

Component Functionality and Quality Testing

In the preliminary testing, the consultants evaluated the MR-Pro and ARTA specifications and functionality in order to determine their utility in the low cost test setup. Initial findings revealed the following insights into the performance of each test.

General Notes

The MR-Pro is a single channel device and only operates at 16-bit and 48kHz.

ARTA is only able to measure one channel at a time.

Frequency Response:

The high level performance guideline states:

> *Frequency response shall be measured at –20 dBFS with a sinewave whose frequency varies from 10 Hz to 50 kHz in steps no larger than 10 per octave.*

The MR-Pro only operates at 48kHz. The frequency response test method and performance specification accommodate both 48kHz and 96kHz sample rates. Naturally, whereas the 96kHz evaluates frequency response from 20Hz to 40kHz, the 48kHz sample rate limits the upper end of the frequency sweep to just over 22kHz.

Otherwise the MR-Pro is able to generate sweep with 12 steps per octave, meeting a portion of the test method. However, a complication emerges in considering the analysis side of the equation. There is not a standardized sweep signal that is used across test and measurement devices. For instance the start and end frequencies, the step size between each transition in the sweep, the duration of each frequency, and the duration of the entire sweep are all different for varying test and measurement systems. On the analysis side this comes into play because the analysis performed needs to accurately detect the frequency, sync up and align with the test signal precisely, and integrate over enough time to accurately measure the signal level, before the signal shifts to a new frequency. Otherwise the results are inaccurate. This makes it difficult to analyze a sweep without a mechanism for synchronization between the generation and analysis sides of the equation.

For the purpose of testing at agencies in the Washington DC area the consultants decided to capture a sweep generated by the AP 2722 and to reproduce this from the MR-Pro. This should have allowed analysis using the AP 2722. However, synchronization was not able to be achieved despite extensive troubleshooting and correspondence with Audio Precision. This ultimately invalidated our efforts on this test and requires evaluation of alternate options.

Total Harmonic Distortion + Noise (THD+N)
The high level performance guideline states:

"Based on AES-17: The EUT shall be stimulated with a low distortion sine wave. The test signal present in the output shall be removed with a notch filter and bandwidth limited from 20 Hz to 20 kHz. The RMS amplitude is reported as a ratio to the RMS amplitude of the unfiltered signal. The measurement should be performed at the following amplitude and frequency combinations: -1.0 dBFS at 41 Hz, 997 Hz and 6597 Hz, –10 dBFS at 997 Hz, and -20 dBFS at 997 Hz, and -60 dBFS at 997 Hz."

For this the MR Pro is able to generate each of the stated frequencies at each of the stated levels. However, for analysis the guideline states:

> *The test signal present in the output shall be removed with a notch filter and bandwidth limited from 20 Hz to 20 kHz. The RMS amplitude is reported as a ratio to the RMS amplitude of the unfiltered signal.*

The ARTA does not provide a means to perform the analysis according to this method. Investigation into other similar software revealed the same issue. ARTA uses an FFT approach, measuring the RMS values of the distortion harmonics. The ARTA user manual explains the methodology it uses and defines the measurement it performs with the following language.

> *THD+N – total harmonic distortion plus noise – defined as percentage of the square root of ratio of power sum of higher harmonics and the noise power to the total signal power that also include distortion and noise power:*

$$THD+N = 100\sqrt{\frac{HarmonicPower + NoisePower}{TotalPower}}\ (\%)$$

Furthermore the ARTA user manual goes on to recognize the methodology referenced in the FADGI guideline and other test methods:

> *In analog instrumentation HarmonicPower+NoisePower is obtained by applying notch filter to the fundamental frequency. The RMS value of signal and signal*

with notched fundamental harmonic are measured in some predefined frequency band, usually from some low frequency cut-off (10, 20 or 100 Hz) to the high-frequency cut-off (22, 30 or 80kHz). ARTA does not use a high frequency limiting. It is automatically done by the antialiasing filter of an input AD converter. The low frequency cut-off can be set by the user.

With all factors in mind the consultants decided to use the ARTA analysis methodology for this test as part of the low-cost test setup. The final reading is reported as a percentage in ARTA at the bottom left of the screen, labeled THD+N.

The test method differs from the one specified in the high level performance guideline, and therefore it does not produce performance metrics that are directly comparable to those generated using the high level performance guideline.

Dynamic Range (Signal to Noise)
The high level performance guideline states:

Based on AES-17: The measurement is the ratio of the full-scale amplitude to the weighted r.m.s. noise and distortion, expressed in dB, in the presence of signal. It includes all harmonic, inharmonic, and noise components. The test signal shall be a 997-Hz sine wave producing -60 dBFS at the EUT output. Any 997-Hz test signal present in the output is removed by means of a standard notch filter. The remaining noise is filtered with an A weighting filter limited to 20 kHz. The results shall be reported as unweighted and A-weighted in dBFS.

The MR-Pro is able to generate a test signal at the stated level and frequency without issue. However, ARTA does not have a specific signal to noise or dynamic range test. The manual provides one thought on achieving a measurement for this, stating:

"If there is no signal at the card input, then RMS shows the input channel S/N ratio."

The consultants felt that this is not a sufficient test method due to the fact that the noise floor of an ADC is apt to change its behavior in the presence of signal. They believe that a better way to do this may be through utilizing the THD+N and THD measures. Subtracting THD from THD+N leaves you with a value for noise only. Doing this using the same 997 Hz signal at -60 dBFS as specified in the high level performance guideline may yield the most meaningful results for the low cost performance guideline.

The test method differs from the one specified in the high level performance guideline, and therefore it does not produce performance metrics that are directly comparable to those generated using the high level performance guideline.

Cross-Talk
The high level performance guideline states:

> *One channel of the EUT is driven with a -1 dBFS sinewave and the maximum amplitude of this frequency appearing in any other channel is noted. The measurement is repeated for each input channel and the maximum amplitude for all channels is determined. This amplitude, expressed in dBFS, is increased by 1 dB and reported. The measurement shall be performed at frequencies of 20 Hz, 1 kHz and 20 kHz.*

The MR-Pro is capable of generating all stated frequency at all stated levels. ARTA is able to perform this measurement as stated. One aspect of the measurement and reporting is that it requires manually placing the cursor on and selecting the relevant frequency in order to get a reading of the level at that frequency.

CMRR
The high level performance guideline states:

> *The input shall be driven from a sinewave generator whose output impedance is less than 100 Ohms. The amplitude is adjusted to achieve -20 dBFS at the EUT output. The signal is removed, and the generator reconnected between the chassis ground and one side of the input. A 600 Ohm resistor is connected between this point and the other side of the input. If the input is asymmetrical, the generator should be connected to the low side and the resistor to the high side. The output should be measured through a bandpass filter at the sinewave frequency. The resulting RMS value, measured in dBFS, is increased by 20 dB and reported as a dB (not dBFS) value. The measurement should be performed at 60 Hz, 1 kHz and 20 kHz.*

There are multiple reasons that led us to the conclusion that this test is not appropriate for the low cost test. The first is that it requires the building of a specialized cable that is not commercially available in any form. One goal of the low cost test is simplicity and the consultants believe that this provides a barrier that will prove to be a real and practical impediment. Furthermore, ARTA is unable to apply a band pass filter for the measurement and reporting of CMRR test results. Finally, there are many devices which do not have an output impedance less than 100 Ohms. For instance, while the MR-Pro does meet this specification, the MR2 does not. The combination of these complicating factors led us to the conclusion that this test should not be included in the low cost test guideline at this time.

Low Frequency Intermodulation Distortion (LF IMD)
The high level performance guideline states:

> *Based on AES-17: IM measurements shall be performed with a twin tone signal with a peak amplitude of -1.0 dBFS. The rms sum of second- and third-order*

difference frequency components in the output are measured and reported in dBFS. The test frequencies shall be 41 Hz and 7993 Hz in a 4:1 amplitude ratio.

The MR Pro does not have the ability to generate the source signal specified in the high level performance guideline. However, the consultants generated and captured this signal using the AP 2722 and the capability of the MR-Pro to playback a WAVE file. ARTA has the ability to measure LF IMD and uses the ratio between the frequencies as a conditional variable to select the measurement method. This is explained in the ARTA manual:

The choice of used method is determined automatically from the ratio of frequencies f_2 and f_1, in the following way:

- If $f_2 / f_1 < 2$ ARTA uses CCIF method and reports difference frequency distortion DFD2 and DFD3 plus IMD (defined with power method).

- If $f_2 / f_1 > 7$ ARTA uses DIN (SMPTE) method and reports modulation distortion: IMDDIN, MD2 and MD3.

- If $2 < f_2 / f_1 < 7$ ARTA uses Power method and reports IMD

This test uses 7993 Hz and 41 Hz as the test frequencies, yielding a ratio of 194.95 and resulting in ARTA's use of the DIN 45403 (SMPTE) method. The manual defines this method as follows:

This method assume that $f_2 >> f_1$, usually f_1= 250Hz, f_2 = 8000Hz in DIN, or f1= 60Hz, f_2 = 7000Hz in SMPTE standard . Amplitude ratio is $I(f_1)$: $I(f_2)$ = 4:1.

The SMPTE measurement method is determined for analog instrumentation. First, the output distorted signal is high-pass filtered at 2000Hz to remove influence of component $I(f_1)$. Then, the filtered signal is amplitude demodulated at frequency f_2, and low pass filtered at 700 Hz to get the power of modulation components at f_2 +/- nf1. Only a few components are used. Finally, IM distortion is expressed as square root of ratio of modulation power to power of $I(f_2)$.

ARTA follows definition from DIN standard called total intermodulation factor:

$$IMD_{DIN} = 100\sqrt{\sum_{n>0} \frac{(I(f_2 + nf_1) + I(f_2 - nf_1))^2}{I^2(f_2)}}$$

In this expression, amplitudes of the sidebands are rms summed and expressed as a percentage of the upper frequency level. This intermodulation factor is very close to the value of intermodulation distortion that can be measured by SMPTE analog instrumentation.

ARTA allows for user defined frequencies and amplitude ratios for the measurement of IMD. Inputting this information is required to perform the tests.

It is worth emphasizing that, as indicated above, the ARTA software application employs an IMD measurement method derived from the relevant ITU-R standard rather than the similar (but not identical) method standardized by the AES and recommended by FADGI in the high performance guideline. (ITU-R methods are sometimes referred to as CCIF, the name of ITU-R's predecessor organization, Le Comité Consultatif International Téléphonique. AES stands for the Audio Engineering Society).

The ARTA tool, however, offers some special settings that may provide a pathway to an AES-compliant result, but there was not sufficient time during the 2015 project to fully test this possibility. The idea is that selecting the 2nd and 3rd order IMD check boxes in the spectrum scaling dialog within ARTA will cause them to be displayed independently of the total IMD value. If this is true then the 2nd and 3rd order IMD values can be identified and an rms sum can be calculated in the AES manner.

High Frequency Intermodulation Distortion (HF IMD)
The high level performance guideline states:

> *Based on AES-17: IM measurements shall be performed with a twin tone signal with a peak amplitude of -1.0 dBFS. The rms sum of second- and third-order difference frequency components in the output are measured and reported in dBFS. The test frequencies shall be 20 kHz and 18 kHz in a 1:1 amplitude ratio.*

The MR Pro does not have the ability to generate the source signal specified in the high level performance guideline. However, the consultants generated and captured this signal using the AP 2722 and used the capability of the MR-Pro to playback a WAVE file. ARTA has the ability to measure HF IMD and uses the ratio between the frequencies as a conditional variable to select the measurement method. This is explained in the ARTA manual:

> *The choice of used method is determined automatically from the ratio of frequencies f_2 and f_1, in the following way:*
>
> *- If $f_2 / f_1 < 2$ ARTA uses CCIF method and reports difference frequency distortion DFD2 and DFD3 plus IMD (defined with power method).*
>
> *- If $f_2 / f_1 > 7$ ARTA uses DIN (SMPTE) method and reports modulation distortion: IMDDIN, MD2 and MD3.*
>
> *- If $2 < f_2 / f_1 < 7$ ARTA uses Power method and reports IMD*

This test uses 20 kHz and 18 kHz as the test frequencies, yielding a ratio of 1.11 and resulting in ARTA's use of the CCIF (ITU-R) method. The manual defines this method as follows:

> *CCIF standard for intermodulation distortion measurements recommends excitation with two closely spaced frequency components $f_2 \sim f_1$. It is recommended to use f_1= 13kHz, f_2 = 14kHz in 15kHz limited system, or f_1= 19kHz, f_2 = 20kHz for amplifier testing. Recommended amplitude ratio is $I(f_1): I(f_2)$ = 1:1.*
>
> *Dominant intermodulation products are at difference frequencies. Second order DFD is at frequency f_2 - f_1., the third order DFDs are at frequencies $2f_2 - f_1$, $2f_1 - f_2$, then follows DFDs at frequencies $3f_2 - 2f_1$, $3f_1 - 2f_2$, … and so on.*
>
> *Many analog instruments that conform to CCIF standard measure only 2nd order difference frequency distortion DFD2, i.e.*
>
> > *IMD_{CCIF} = DFD2 (in analog instrumentation)*
>
> *Some CCIF instruments also measure 3rd order difference frequency distortion DFD3.*
>
> *Due to the close frequency separation, this technique is also applied in some swept-frequency analyzers.*
>
> *Modern FFT analyzers are capable of measuring all distortion products. ARTA reports DFD2 and DFD3 and also a total intermodulation distortion (IMD), calculated by power method using twenty strongest intermodulation spectrum components.*

ARTA allows for user defined frequencies and amplitude ratios for the measurement of IMD. Inputting this information is required to perform the tests.

It is worth emphasizing that, as indicated above, the ARTA software application employs an IMD measurement method derived from the relevant ITU-R standard rather than the similar (but not identical) method standardized by the AES and recommended by FADGI in the high performance guideline. (ITU-R methods are sometimes referred to as CCIF, the name of ITU-R's predecessor organization, Le Comité Consultatif International Téléphonique. AES stands for the Audio Engineering Society).

The ARTA tool, however, offers some special settings that may provide a pathway to an AES-compliant result, but there was not sufficient time during the 2015 project to fully test this possibility. The idea is that selecting the 2nd and 3rd order IMD check boxes in the spectrum scaling dialog within ARTA will cause them to be displayed independently of the total IMD value. If this is true then the 2nd and 3rd order IMD values can be identified and an rms sum can be calculated in the AES manner.

Amplitude Linearity
The high level performance guideline states:

> *Based on AES-17: Level-dependent logarithmic gain is measured at 997 Hz from -5 dBFS to -105 dBFS and reported as standard deviation value in dB.*

The MR-Pro does not have the ability to generate this level sweep. The unit is also 16-bit, with dynamic range more limited than the level sweep specified. Finally, there is no test for this in ARTA or similar software. For these reasons the consultants believe that amplitude linearity is not appropriate for the low cost test guideline.

Spurious Aharmonic Signals
The high level performance guideline states:

> *A 997 Hz sinewave shall be applied at -1 dBFS. The output spectrum shall be measued with an 32,768 point FFT. The largest inharmonic component is reported in dBFS.*

The MR-Pro is capable of generating a signal at the stated frequency and level. ARTA is able to measure using a 32k point FFT. Measuring and reporting the largest aharmonic component requires manual review and selecting the frequency where the largest component appears in order to get a readout from ARTA.

Alias Rejection
The high level performance guideline states:

> *Based on AES-17 and IEC 61606-3: The device is stimulated with a variable frequency sine wave at -10 dBFS. Beginning at half the sample rate, the frequency is continuously increased until it reaches 200 kHz. For a 48 kHz sample rate, the frequency is swept from 24 kHz to 200 kHz. For a 96 kHz sample rate, the frequency is swept from 48 kHz to 200 kHz. The rms amplitude at the converter output, increased by 10 dB, is graphed. Results are reported as the lowest frequency at which the alias component was equal to or greater in amplitude than all other alias components across the frequency range tested. Amplitude is expressed relative to the stimulus amplitude in dB.*

The 48kHz limitation of the MR-Pro limits the upper end of its frequency range to a maximum of 24kHz, making this test unable to be performed using these components. This test is not appropriate for the low cost performance test.

Sync Input Jitter Susceptibility and Jitter Transfer Gain
Both of these tests call for driving the ADC reference input with a jittered signal. The MR-Pro does not have this capability. These tests are not appropriate for the low cost performance test.

Basic System Setup
Based on our findings above the consultants created a series of presets and transferred files that were generated using the AP2722 to the MR-Pro. The details of this are identified in the test method. Because the device is a single channel unit they also created a high quality cable, splitting the signal output from the MR-Pro. Special attention must be paid to the quality and proper functioning of such a cable, and this should be noted by those performing tests using the same methodology. Low quality cables will degrade the signal and impact the tests significantly.

The consultants installed ARTA on a Windows laptop running Windows 7. This laptop also contained Sequoia recording and editing software which was used to capture the output of the ADC. The interface used for capturing the digital output of the ADC was a Sound Devices USBPre2 interface, designed to accept both SPDIF and, via an adapter, AES digital inputs.

Signals were generated using the MR-Pro, routed through the ADC, and captured to the Windows 7 laptop via the Sound Devic2 and Sequoia software. Figure 1 illustrates the setup.

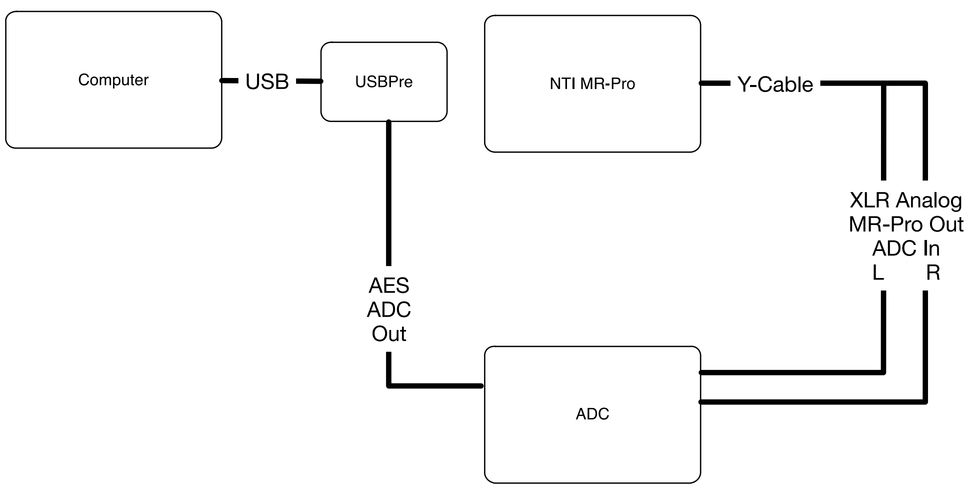


Figure 5: Basic Diagram of Test Setup

This diagram demonstrates how the signals are captured, but does not explain how the captured signals are analyzed. ARTA and applications like it do not analyze files in non-real-time. They expect a real-time signal input to analyze. In other words, you can not select the test you want to perform, select the files for testing, and select "analyze". You have to playback the recorded files into ARTA and analyze them as if you were analyzing them in real-time. As previously mentioned the consultants used one laptop containing both Sequoia and ARTA software. To analyze the signals they used a virtual cable utility (VB-Audio Virtual Cable) to route the signal from the output of Sequoia into ARTA. Once this signal path was established analysis was performed by playing back the signal from Sequoia in real-time and analyzing within ARTA. In this way the consultants were able to separate the processes of signal generation and capture from analysis. This is advantageous in meeting the original vision of a service provider returning recorded test signals back to a client for testing.

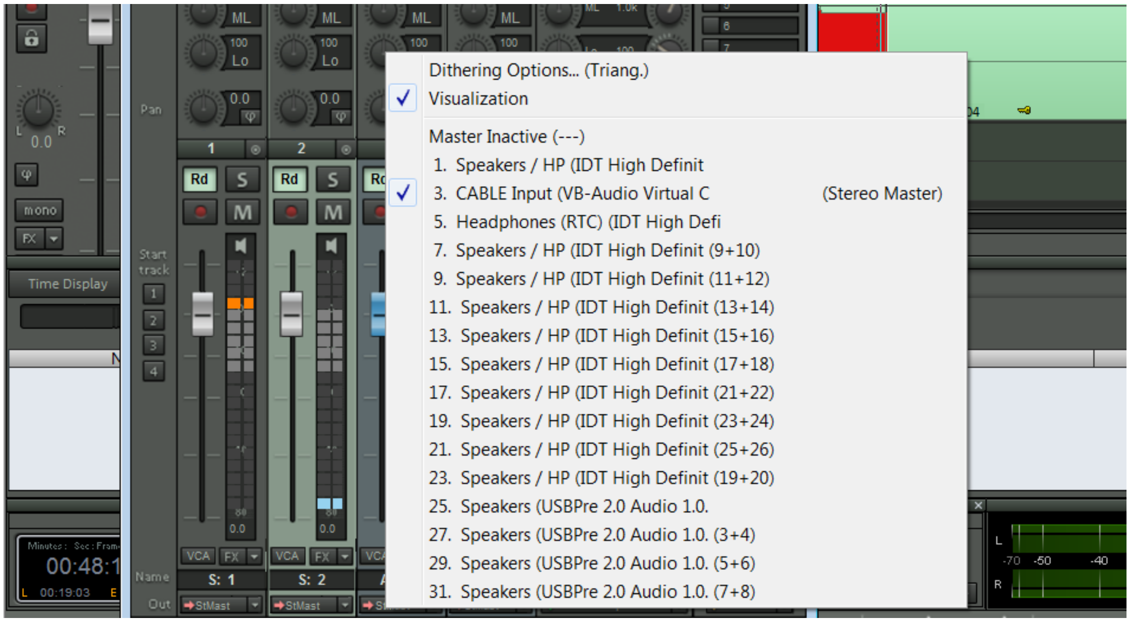


Figure 6: Demonstrating selection of outputs from Sequoia where signals are played back

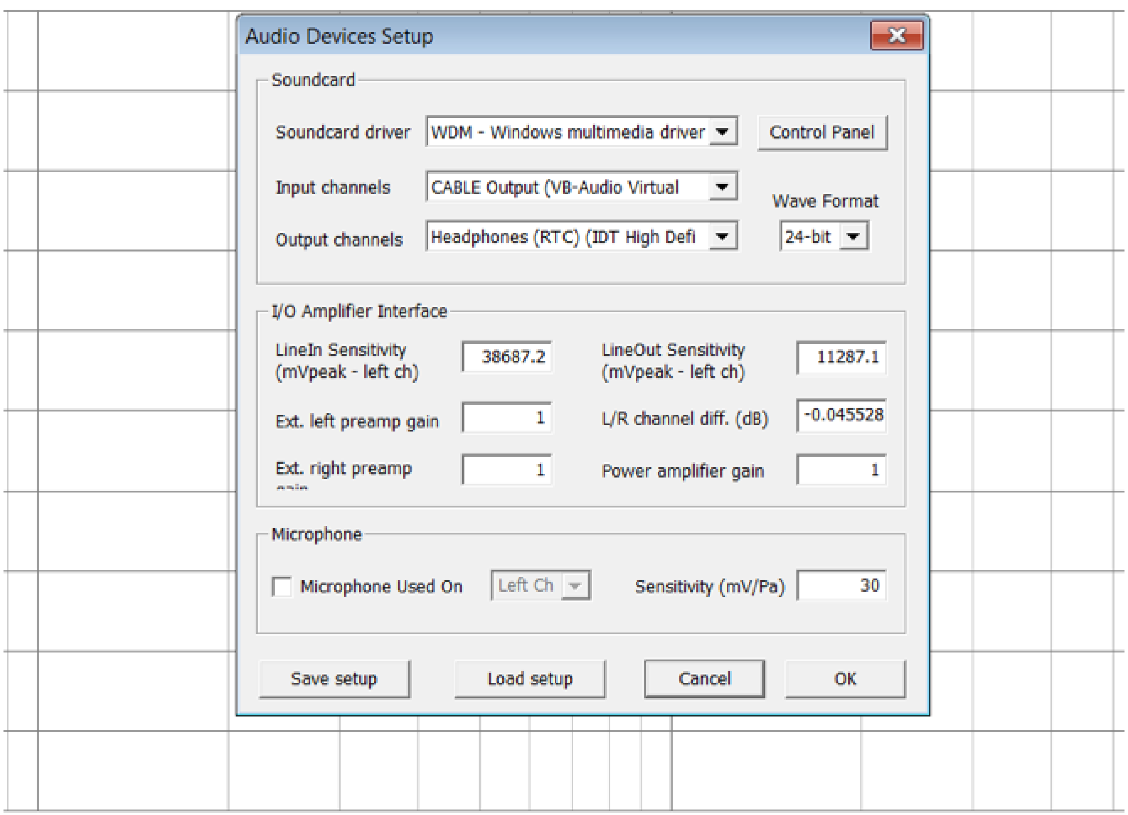


Figure 7: Demonstrating selection of inputs within ARTA where signals are analyzed

All reporting is performed by manually reviewing results within ARTA and documenting them. There is no automated reporting or exporting of results as tabular data.

Challenges and Takeaways

Assumptions

There were two core assumptions that turned out to be inaccurate.

1. The consultants originally anticipated that generating all of the test signals using the AP 2722, recording them and placing them on the MR-Pro, and having the MR-Pro serve as the playback device would yield the best results. This turned out to be incorrect. The signals that could be generated by the MR-Pro turned out to be significantly higher quality than the same signals generated by the AP 2722 and played back by the MR-Pro. However, there are some signals that could not be generated by the MR-Pro, such as the IMD test signals. For these they found that the highest quality was achieved when the signals were captured at a level which would allow the MR-Pro to operate at its own maximum level. In other words, for a source signal specified at -20dBFS, better performance was achieved when the MR-Pro output a -20dBFS signal while set to maximum level output. This is in contrast to loading a file which would require a level reduction on the MR-Pro in order to achieve -20dBFS.

2. In following the lead of the FADGI still image working group, the consultants originally anticipated that the tests for moderate quality performance would be a subset of those used for the high performance tests, and with less stringent performance metrics compared to the high performance tests. However, the “hard wired” test methods used by ARTA and similar software may vary from those defined in the FADGI high performance test guideline. Where this is true this complicates matters further because it disallows direct comparison of performance metrics generated using the high performance test method and those generated using the low cost test method. This could lead one to argue that the language of the high level, performance guideline is biased in favor of the measurement methodologies incorporated into the AP 2722, specifically in this case where the choice of the 'standard notch filter' is favored over the other methodologies for the purpose of creating consistent results between the analog and digital analyzers built in to the AP. Further testing is needed to compare the results of the AP 2722 and of ARTA for the same source signals.

MR-Pro vs ARTA

ARTA does have its own signal generator which brings up the question of whether or not the MR-Pro should be eliminated and replaced by the ARTA altogether. There are a couple of reasons the consultants think that this is not the best approach. If the ARTA is used alone then a Digital-to-Analog-Converter (DAC) must be used, and the converter will have its own performance limitations that will impact the test. It is possible to measure and compensate for the DAC limitations in the testing and reporting. However, if this is not necessary then they feel that it’s best to leverage the consistency of the

MR-Pro where possible and avoid the variables that come along with utilizing different DACs and proposing an additional calibration step into the test method in order to perform DAC compensation.

Unit of Measure
Many of the test results in the high performance guideline require reporting in dB. ARTA reports many of its results as a percentage. This requires performing a conversion from percentage to dB. The equation is (LOG10(%/100))*20. For instance, if the ARTA value is 0.01% then the equation would be (LOG10(.01/100)*20 = -80dB

Calibration
The AP 2722 and similar devices have the ability to self regulate and calibrate to ensure proper level prior to running a given test. Calibration using this more manual approach, and with more variables may create some challenges for users. Additional time will need to be put into figuring out a way of addressing this challenge.

IV. Proposed Test Method for the Partial Minimum Metrics, Low Cost System

The test method can be divided into two main portions. The first is the signal generation and the second is the analysis.

Signal Generation and Capture

Initial Calibration
Load Recall *Config 0* (997Hz at 18dBu)
Change level on Minirator until you see -1dBFS on the meters of the system being recorded to
Document the level on the MR-Pro that produces -1dBFS

Frequency Response (20 – 20kHz generated using AP2722 in compliance with specification)
Load *File 1* (begins with 1kHz tone at beginning for calibration, followed by frequency sweep)
Set level on MR-Pro to -20dBFS using 1kHz tone
Hit record in capture software
Hit play on the MR-Pro *file 1* from beginning
Record until it goes through the sweep and back to the 1kHz, capturing 1Khz before and after the sweep in the file.

THD+N
Load *Config 1* (41Hz)
Set level to level that = -1dBFS (17dBu)
Record for 10 seconds or so

Load *Config 0* (997Hz)
Record for 10 seconds or so
Set level of MR-Pro to -1dBFS within capture software

Load *Config 0* (997Hz)
Set level of MR-Pro to -10dBFS within capture software
Record for 10 seconds

Change level of MR-Pro to -20dBFS within capture software
Record for 10 seconds

Change level of MR-Pro to -60 dBFS within capture software
Record for 10 seconds

Load *Config 2* (6597Hz)
Set level of MR-Pro to to -1dBFS within capture software
Record for 10 seconds

Dynamic Range
Load *Config 0*
Set level of MR-Pro to -60 dBFS within capture software
Record for 10 seconds

Cross Talk
Load *Config 3* (20Hz)
Set level of MR-Pro to -1dBFS within capture software
Short channel 1 using the shorting plug (mimicking output impedance of device)
Record for 10 seconds
Short channel 2 using the shorting plug (mimicking output impedance of device)
Record for 10 seconds

Load *Config 0* (997Hz)
Set level of MR-Pro to to -1dBFS within capture software
Short channel 1 using the shorting plug (mimicking output impedance of device)
Record for 10 seconds
Short channel 2 using the shorting plug (mimicking output impedance of device)
Record for 10 seconds

Load *Config 4* (20kHz)
Set level of MR-Pro to to -1dBFS within capture software
Short channel 1 using the shorting plug (mimicking output impedance of device)
Record for 10 seconds
Short channel 2 using the shorting plug (mimicking output impedance of device)
Record for 10 seconds

IMD LF (need a peak reading meter. Not an RMS meter for this test to measure the -1dBFS)
Load *File 2 (Twin tone test signal compliant with high level guideline generated by AP2722)*
Set level of MR-Pro to to -1dBFS within capture software
Record for 10 seconds

IMD HF (need a peak reading meter. Not an RMS meter for this test to measure the -1dBFS)
Load *File 3 (Twin tone test signal compliant with high level guideline generated by AP2722)*
Set level of MR-Pro to to -1dBFS within capture software
Record for 10 seconds

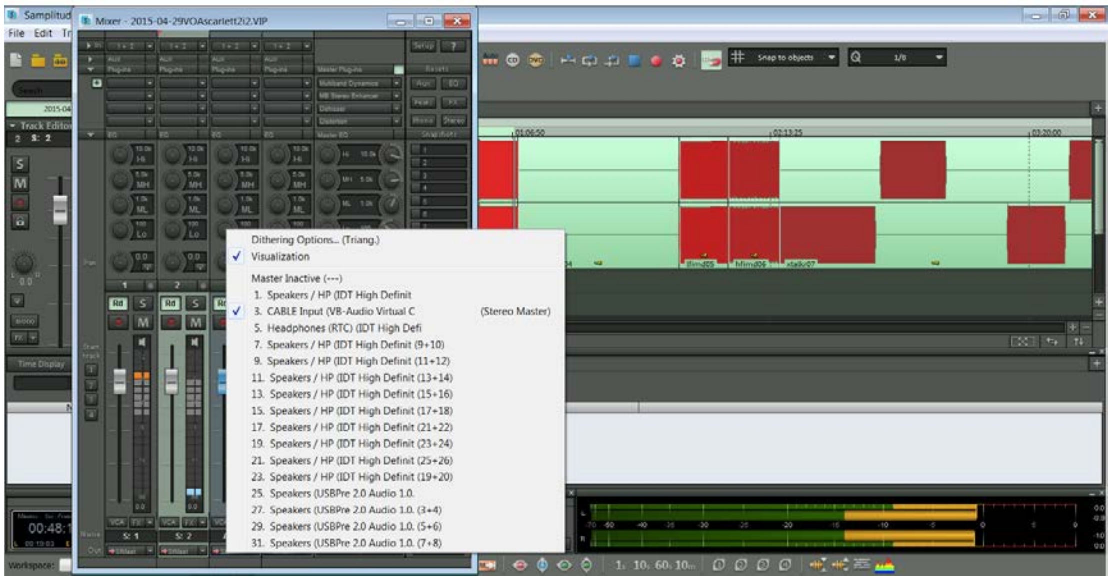
Spurious Aharmonic Signals
Load *Config 0* (997Hz)
Set level of MR-Pro to to -1dBFS within capture software
Record for 10 seconds

Analysis

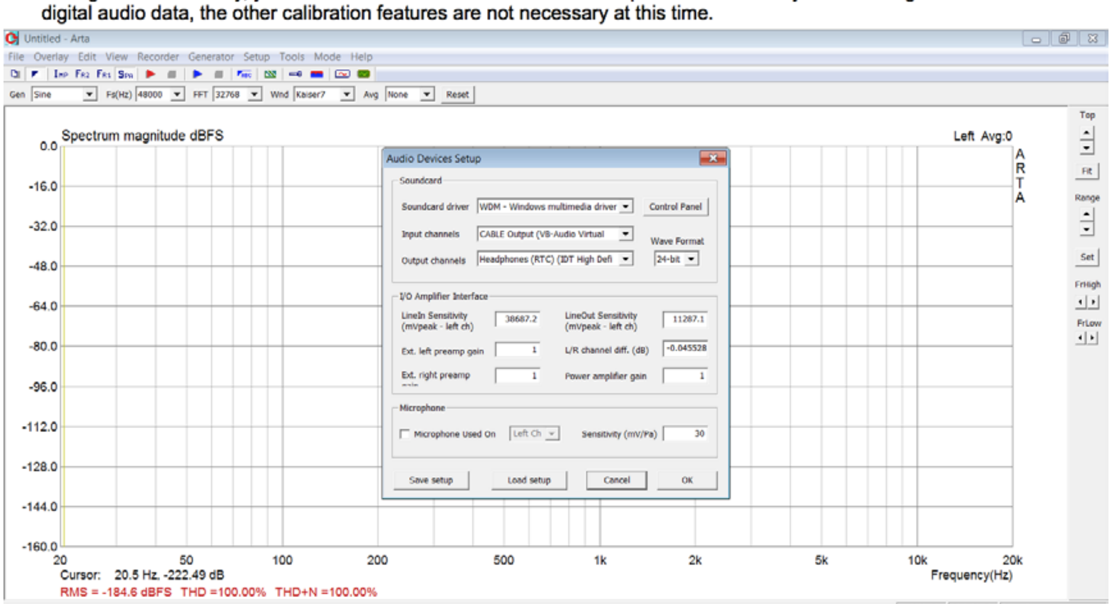
Due to the manual nature of the analysis process the following test method is presented in the format of a tutorial for maximum understanding.

For analyzing recorded files

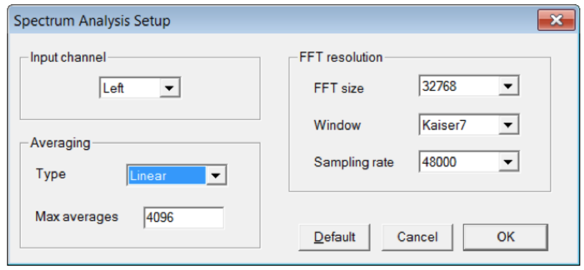
To route the audio from the DAW to ARTA I installed a WDM driver called VB-Audio Virtual Cable which is donation-ware. There is also an ASIO version available.



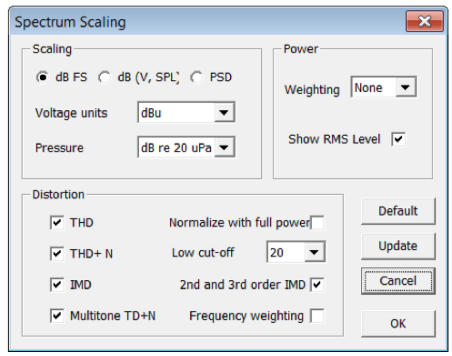
To get started in ARTA, Select Setup>Audio Devices and in the 'Input Channels' box select Cable Output (VB-Audio Virtual Cable) to route the audio from the DAW into Arta. If you are using an audio interface and testing the ADC directly, you would choose the relevant driver and input. Since the analyzer is looking at digital audio data, the other calibration features are not necessary at this time.



Under menu: Setup>measurement, you get the spectrum analysis setup menu. I recommend the settings shown as a good starting point for the measurements undertaken. Make certain that the sample rate chosen matches that of the Device Under Test. You also use this page to switch between left and right channels



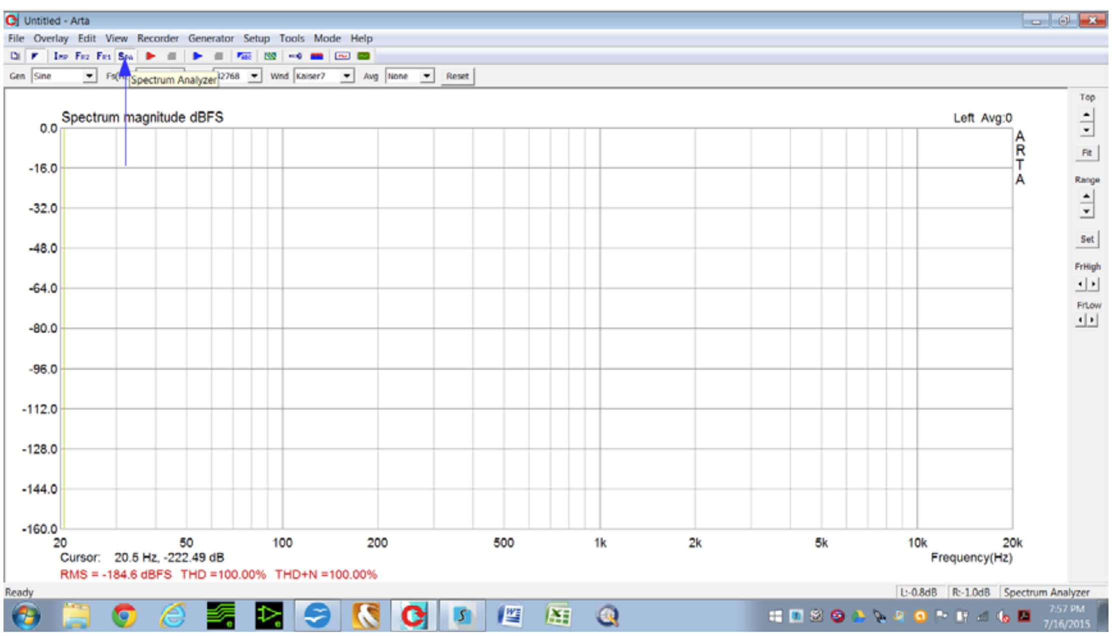
Next, under Setup> spectrum scaling, select the scaling to dBFS and voltage units to dBu. Weighting: None for most measurements, (some call for A weighting scale) and while you're here you might as well select THD, THD+N and both IMD and 2nd and 3rd order IMD. You'll need those.



THD+N

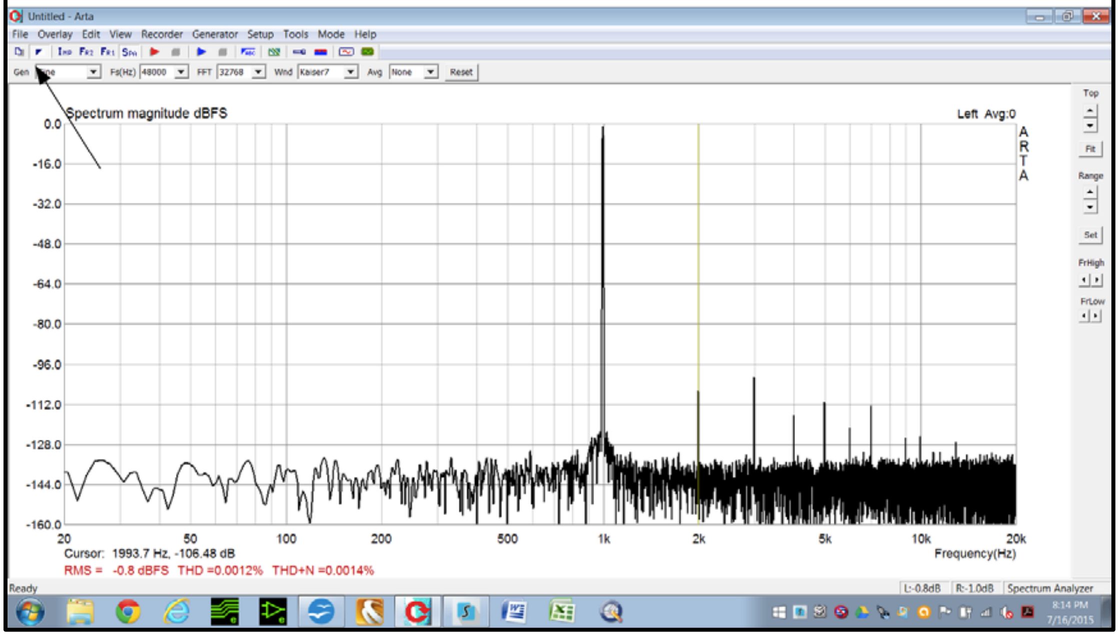
For most of the analysis measurements, select the Spectrum Analyzer button. For THD+N measurements, the 'Gen' pulldown menu should be selected to Sine.

Next, select the section of audio to analyze in the DAW, (I usually loop it) and hit the Red Triangle under the Recorder menu to start the Arta FFT acquiring data. Let the audio play until it is settled, then hit the square box to freeze it to read the data.

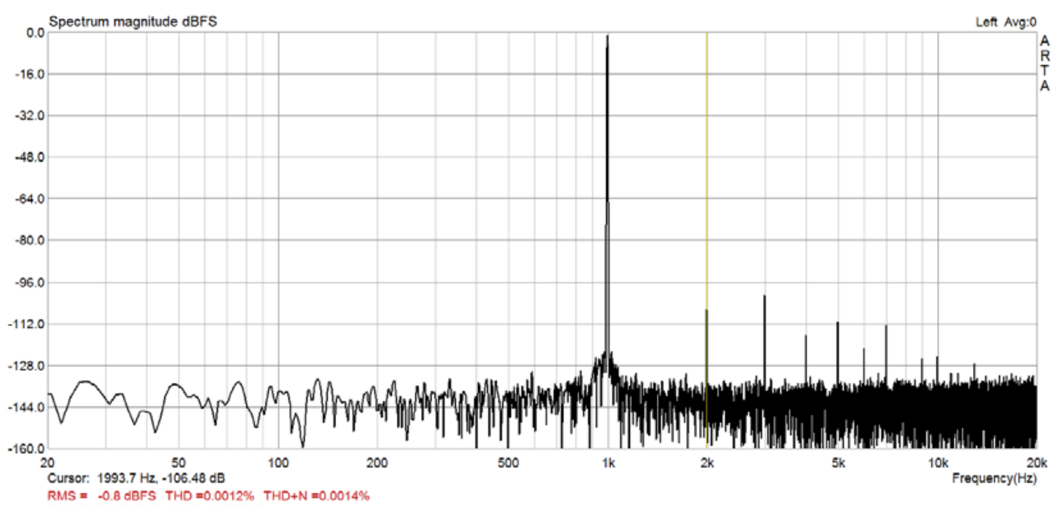


Left click on the screen and either drag or use the arrow keys to set the cursor to an area of interest for measurement: in this case the second harmonic.

You can copy the screen to memory for pasting into another application, and add titles by selecting the copy icon in the upper left corner of the display.

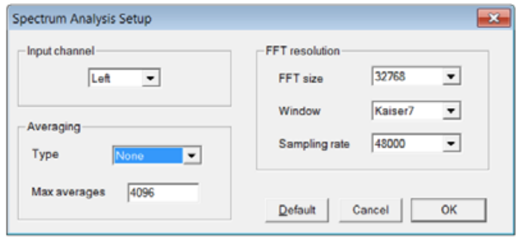


An example of spectrum captured using the technique just described.

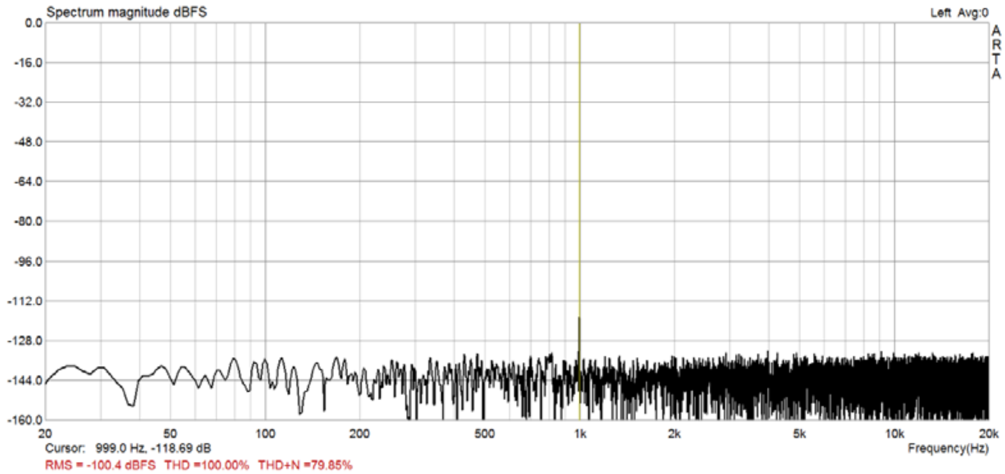


Crosstalk

For Crosstalk, the same spectral analysis setup as used for THD is used, being certain to choose the channel that is not driven with the stimulus for measurement under the Setup>measurement settings.



Run the FFT to acquire the data for measurement. (Red Triangle Icon) Freeze the FFT by hitting the red square 'stop' icon under the record menu. Then left-click on the menu and use the arrow keys to move the cursor to the point of measurement. In this case the stimulus was 997Hz at -1dBFS on the right channel, so cursor to the area around 997 Hz, and select the maximum point for measurement.



EXAMPLE OF CROSSTALK MEASUREMT WITH 997Hz STIMULUS. THE MAXIMA WAS MESURED AT 999HZ,

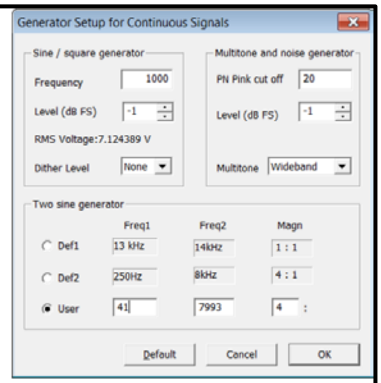
LF IMD

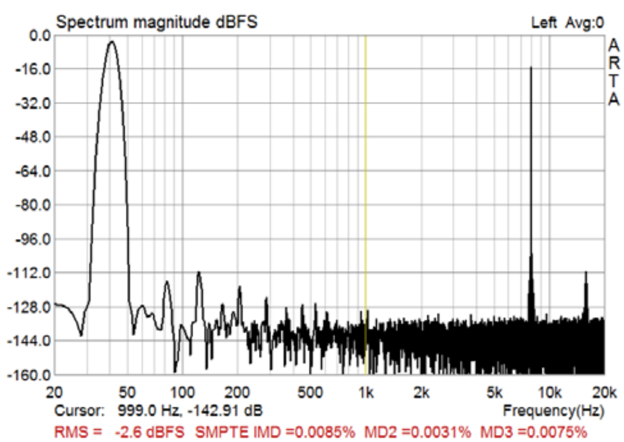
For Low Frequency inter-modulation distortion measurements select Generator>Configure to display the setup menu for the generator. At the bottom, choose the user button in the two-sine generator area and enter the frequencies and ratio you want to analyze.

After hitting OK, select the two-sine option under the 'gen' pull-down menu in the spectrum analyzer.

Play the tone to be measuered, (from the DAW or other source) run the FFT, and read the results.

Make sure that the IMD and 2[nd] and 3[rd] order IMD measurement options have been selected in the Setup>measurement settings window as mentioned earlier.

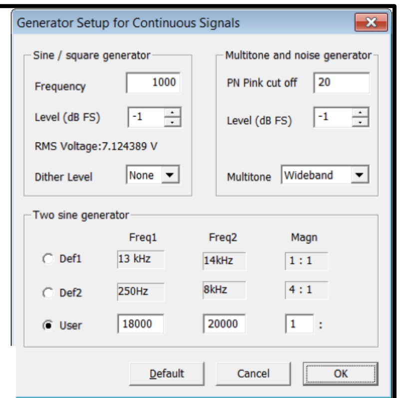




LOW FREQUENCY (SMPTE) INTERMODULATION DISTORTION

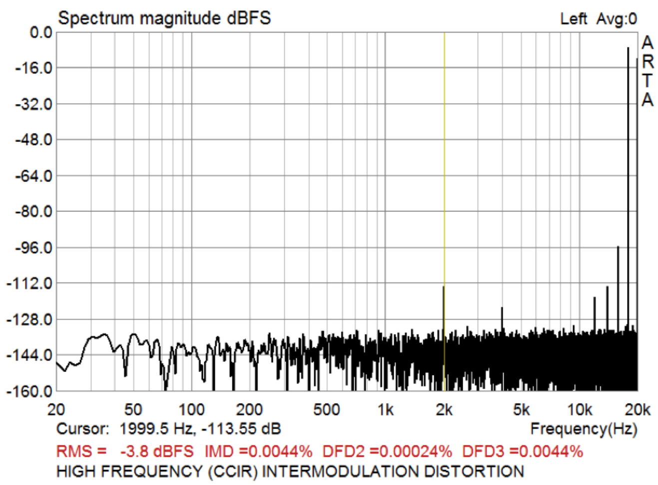
HF IMD

For High Frequency inter-modulation distortion measurements the setup is very similar. Choose Generator>Configure to display the setup menu for the generator. At the bottom, choose the user button in the two-sine generator area and enter the frequencies and ratio you want to analyze. 1:1 ratio in this case.



With the two-sine option still selected under the 'gen' pull-down menu in the spectrum analyzer, Play the tone to be measured, run the FFT, and read the results.

I should note, that configuring the generator sets up the analyzer to measure the results for those frequencies, whether or not you actually use the tones generated in Arta, or in another tone source, such as .wav files played from another source such as the MR-Pro, as done for this initial testing. The important thing here is that the Arta is clocked from the source – either the daw or the ADC under test.



Many of the performance specifications from the FADGI guideline are stated in decibels , where as most audio analyzers report the ratio of distortion readings as percentages.

To convert from percent to dB: $dB = 20\log_{10}(percent/100)$

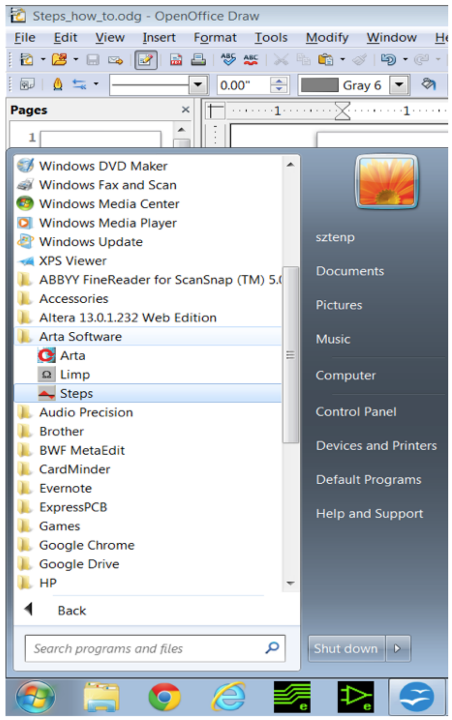
Percent is a ratio in this case, the ratio of the measured distortion artifacts + noise to the stimulus signal.

To convert from dB to percent: $10^{\left(\frac{dB}{20}\right)} x\,100 = percent$

..

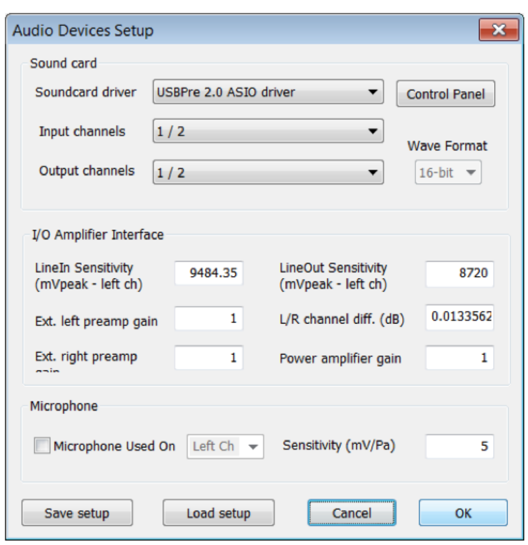
For measuring frequency response in real time using ARTA

Using STEPS to measure the frequency response characteristics of an A to D converter.

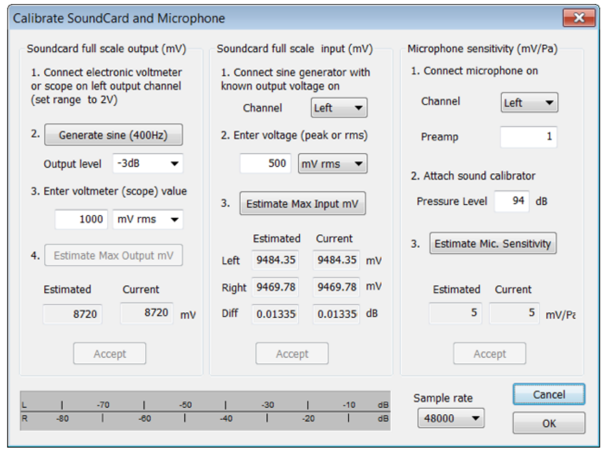


After launching the program, select Setup>Devices to choose the Soundcard driver and relevant inputs for your device. Also, open the control panel and select the correct sample rate, and clocking source to sync with the digital output from the device under test.

If this is your first time using Arta or Steps, refer to the user manual for the calibration procedure.



Under Setup>calibrate devices you will find the panel that allows one to calibrate the inputs and outputs of Steps, and Arta.

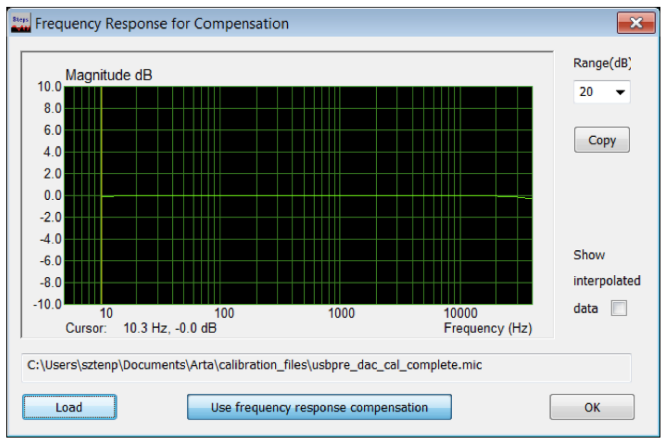


In this case, we are only interested in the output calibration since the measurements of the input signal will be digital which are inherantly calibrated.

The panel walks you through the process. You will need a voltmeter with true RMS reading capability for this step.

Next, you will need to meaure the frequency response of the audio device D-to-A in order to generate a calibration file to eliminate its non-linearity from the measured response of the device under test. For information on how to do this, refer to the user manual

The calibration file is a simple ascii text file, renamed with a xxx.mic extension (instead of xxx.txt). To instantiate it into the measurement process, select the menu under Setup> Frequency compensation, and then hit the load button to navigate to where your calibration file is stored. Once it's successfully loaded, you will see the compensation curve in the graph. Engage the frequency compensation curve by hitting the' use freqency response compensation' button.



Example Frequency Response Compensation File.

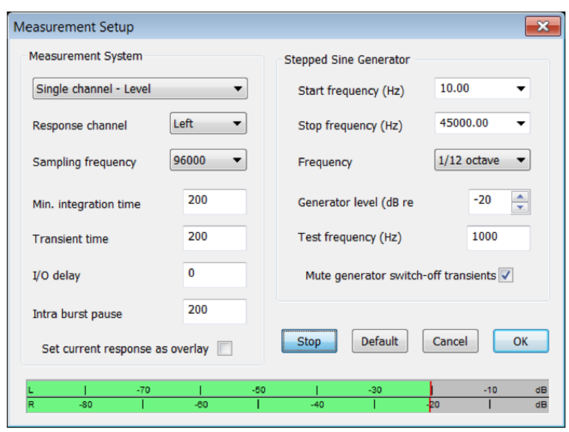
X (Hz)	Y (dBu)
10.25	-0.04765086
10.99	-0.04715086
12.45	-0.04605086
13.92	-0.03585569
15.43	-0.02924782
16.86	-0.02505063
18.33	-0.02217297
20.54	-0.02031567
22.01	-0.01958460
23.53	-0.01924477
24.93	-0.01491483

....

37512.58	-0.28082721
39743.51	-0.30780740
42107	-0.34580669
44611.1	-0.41445014
47263.89	-0.50966816
50074.16	-0.60033079

Having established the calibration of the audio interface, we're almost ready to measure.

Open the Mesurement Setup dialog under the menu Setup>Measurements and configure it for Single Channel-Level. Select which channel you want, and the Sample Rate of the Device Under Test.



(Dual Channel uses one channel as a reference against which the second channel is measured – which is not wanted in this case.)

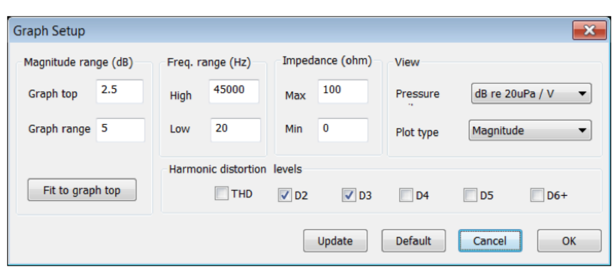
Enter the Start and Stop Frequency for the sweep, the upper limit of which is dependant on the sample rate chosen and cannot be more the sample rate/2.1.

Next you can select how many steps per octave. 1/12 octave (per step) means there will be 12 steps measured per octave.

Set the generator level for the desired output.

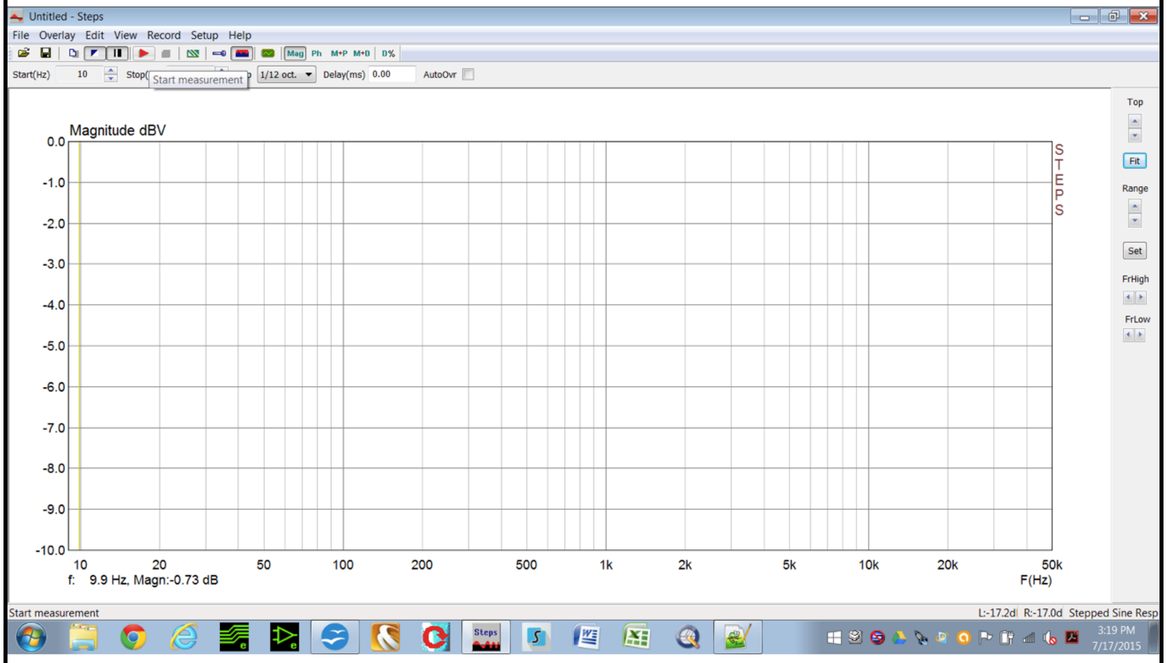
Then select the generate button to test the level.

If you want, you can pre-scale the graph by selecting the Graph Setup dialog under Setup>Graph Here you can set up the vertical and horizontal axiis to be displayed,

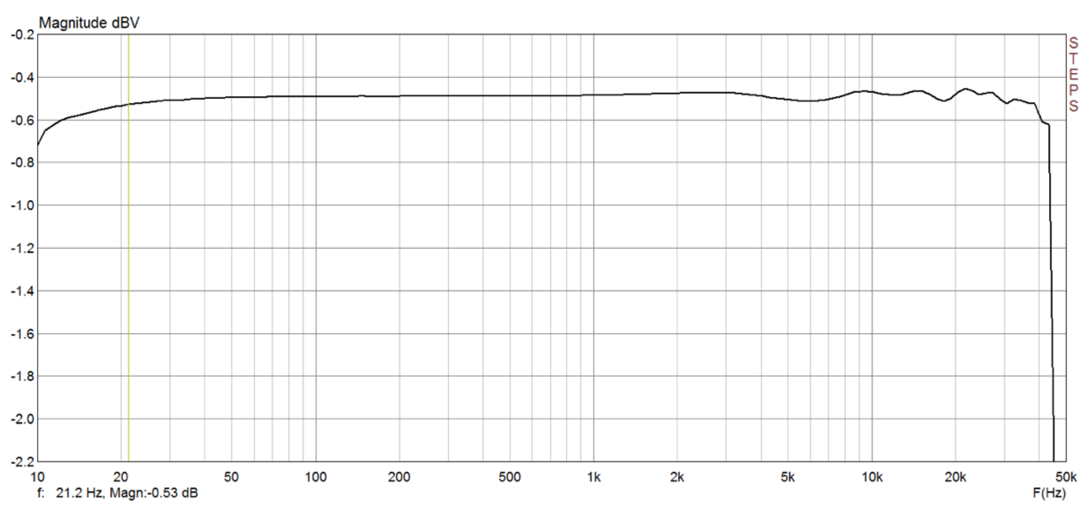


These settings can also be adjusted later using the Top, Range, FR High and FR Low buttons in the main window.

Now you are ready to run the frequency sweep by hitting the red triangular icon to 'start measurement'. You should see the measurements start to be read out in the bottom left corner of the screen: as in this example f 9.9Hz Magn(itude) 0.73dB.

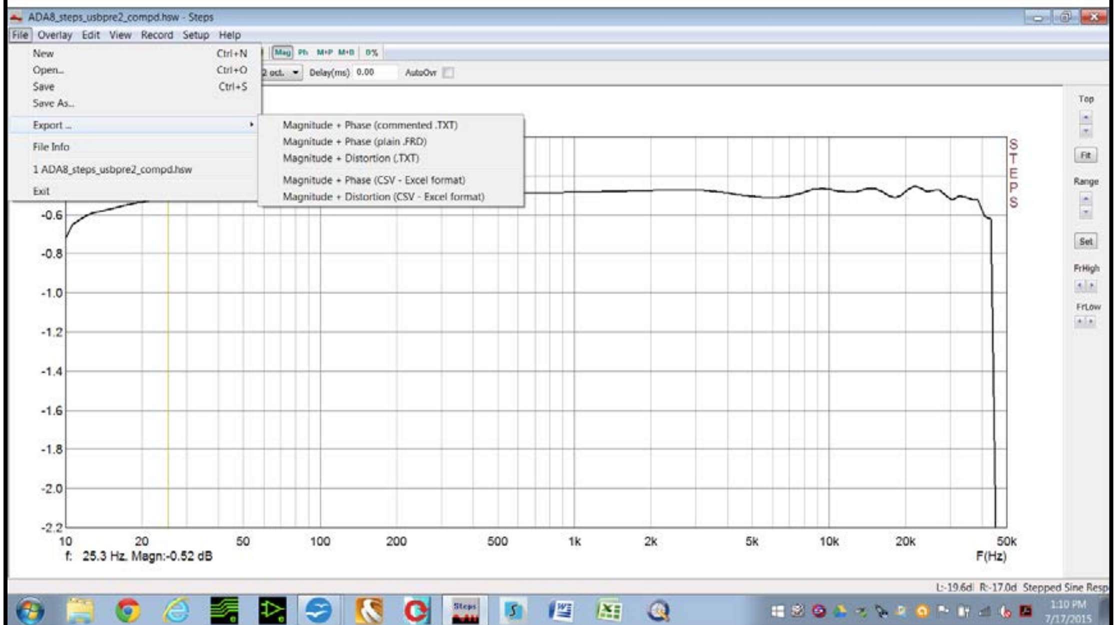


As in Arta, graphic data can be copied and pasted into another program like Open Office for reporting and documentation of results.

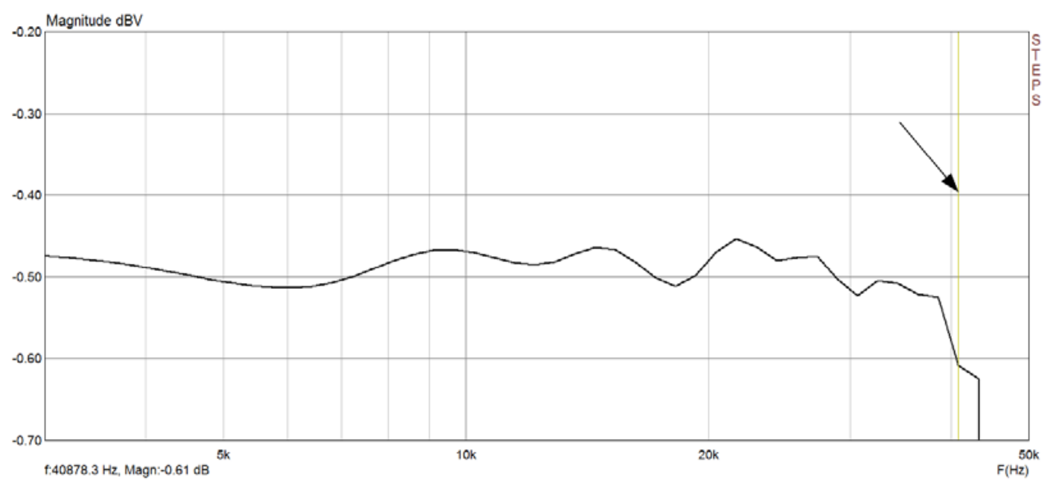


ADA8 Frequency sweep from 10Hz to 45KHz using USB Pre 2 with frequency compensated output

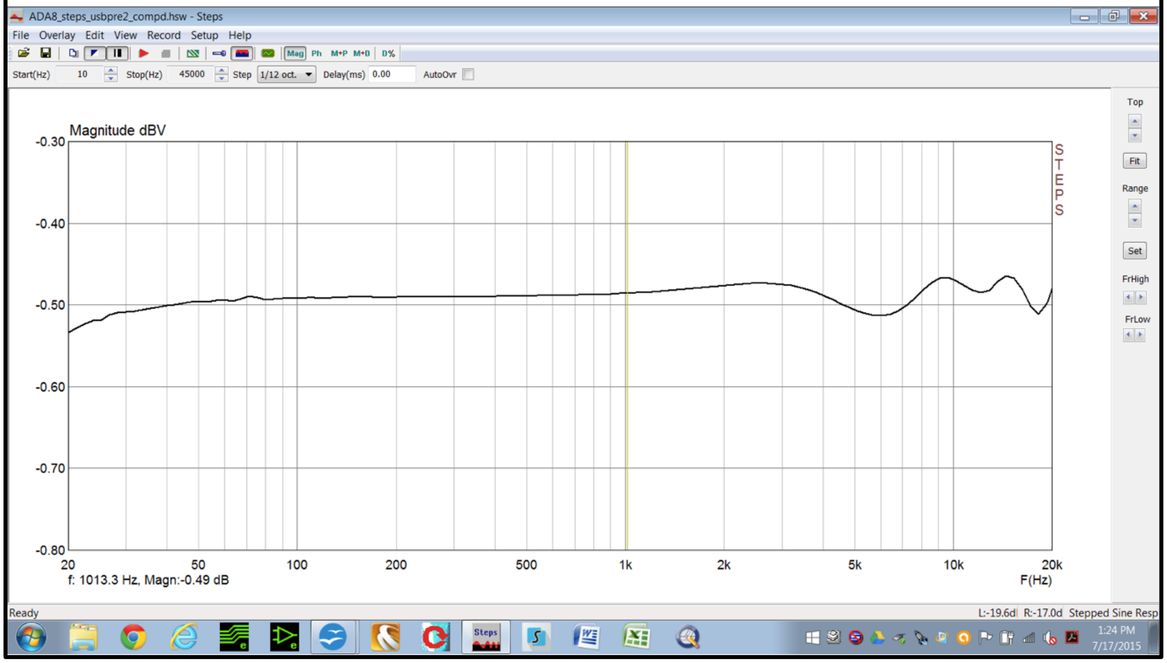
Also tabular data can be exported in a number of formats for detailed documentation or further analysis.



Also, the cursor can be moved to any point on the graph and the magnitude can be read for that point



By using the range and FR High and FR Low buttons, graphic representation of the sweep can be scaled to clearly indicate a pass/fail condition in the Frequency Response in accordance with the standard.



NTI MR-Pro Analysis

The following table reports on the specifications of the NTI MR-Pro as tested and reported by the AP 2722.

MR-PRO GENERATOR ANALYZED W/ AP2722			
Test			**comments**
THD+N	**dBr**	**Percent**	
41Hz	-95.4	0.00170%	
997Hz @ -1dBFS	-95.7	0.00164%	
997Hz @ -10dBFS	-94.6	0.00186%	
997Hz @ -20dBFS	-91	0.00282%	
997Hz @ -60dBFS	-51.7	0.26002%	
6597Hz	-96.2	0.00155%	
Dynamic Range			
Unweighted	-93.2		
A-weighted	-96.8		
Spurious Aharmonic Signals	-133.9		@3353Hz
LF (SMPTE) IMD	-81.5	0.00841%	
HF (CCIR) IMD	-77.4	0.01349%	

Operator Skill Level Requirement

The execution of an ADC test using the partial minimum metrics, low cost system requires the operator to have a moderate-level knowledge of audio engineering and digital technology. A sense of this requirement is provided in section V, in the description of the field testing, which includes a step-by-step script to guide the operator carrying out the test. To a degree, some parts of the path through this script can be made easier by the provision of software support. However, the intricacies of the test and the vagaries of operating the analyzer and ADC devices (each with quirks, as described in section V) mean that this test will never be simple enough for a lay person to carry out.

V. Testing ADCs in Federal Agencies

Purpose for the activity

The purpose of the testing was to verify that the approaches and tools developed at the consultant facility would work in the field, and to identify aspects that require adjustment. The entire process also provided valuable information that the consultants used to draft recommended changes to the high level ADC performance guideline, and to compile an initial recommendation for low-cost guideline.

In December 2014, in a meeting of the FADGI Audio-Visual Working Group, member agencies were asked if they would be interested in hosting on-site testing. Four agencies responded in the affirmative:

- National Archives and Records Administration (NARA)
- Voice of America (VOA)
- Smithsonian Institution Center for Folklife and Cultural Heritage
- Library of Congress Packard Campus (LC)

The testing visits were set up for a two-day span in April 2015. A variety of scheduling and logistical factors limited this round of testing to NARA, VOA, and LC. Prior to the visit, the expert consultants corresponded with each organization in order to identify the devices being tested and their associated audio interface options. In each case suitable high-end ADCs were identified and, at VOA, an additional low-cost ADC was provided for testing. This exercise was designed to assess the performance-testing method, and did not include the sample-selection and statistical features that would be required for bona fide comparison testing of ADCs. For this reason, this report does not identify the specific ADCs that were tested. They are referred to as NARA ADC, VOA high-end ADC, VOA low-end ADC, and LC ADC.

LC staffers and FADGI coordinators Carl Fleischhauer and Kate Murray aided in arranging the meetings and accompanied Chris Lacinak and Phillip Sztenderowicz to each site visit.

High performance test procedure

General comments

While test files and macros were created for each individual high performance test, at the time of the visit the consultants did not have one master macro that would run all tests. Therefore it was important to create a test procedure outlining the exact order and protocol for the tests. This ensured that all tests were performed, and performed accurately. On the reporting side, there are multiple ways that data can be captured and presented within the Audio Precision 2722. There are two main types of test results that are generated. The first is one that contains raw data and accompanying graphic data. To capture the raw data, the test must be setup to output to the AP 2722 log, and logging must be turned on. This will capture the test that was run, along with the date and time that it was performed, whether the test passed or failed (if applicable), and each of the data points captured in the analysis. For certain tests, such as frequency response, the AP 2722 will also present a graph showing the pass and fail threshold lines and the recorded response of the device under test. The second main type of test result contains the raw data and tabular data. In this case the same data is output to the log file. However, instead of a graph the visual display of the result is in tabular form. An example of this is with THD+N where the value reported is a single value for each channel.

As mentioned the log data must be setup to record properly. This had to be done for each test, and the log file had to be saved and then cleared after the testing of each device in order to maintain alignment between results and devices. In addition to this the consultants captured the graph data in order to present the results visually as well. This had to be done for each test by performing a screen grab and saving it to the appropriate directory so that it could be identified at a later date.

Test procedure

Make sure that the ADC is set to its internal clock instead of clocking off of the test device.

Make Sure Data Logging is turned on.

Create Directory for name of Org and Date.

Frequency Response

- Open Frequency Response Test Preset File for 20Hz – 20KHz
- Select Regulate (sets the right level)
- Select Go on the sweep button

- Open Frequency Response Test Preset File for 20kHz – 40KHz
- Select Regulate (sets the right level)
- Select Go on the sweep button

- See graph/log file for results

- Save Graph to directory

Dynamic Range

- Open the Dynamic Range Macro (it will open the dynamic range preset file for you)

- Select the Run Macro button (Looks like a play button)

- The Data Editor Table will show the A-Weighted result. See the log for the unweighted and A-Weighted Results

- No graph

THD+N

- Open THD+N Test Preset File Part 1 (3 frequencies at -1dB)
- Select Regulate button to regulate input
- Select go on the sweep dialog

- See Tabular data (no graph)

- Open THD+N Test Preset File Part 2 (997 Hz at 3 different levels)
- Select go on the sweep dialog

- See Tabular data (no graph)

IMD LF (SMPTE)

- Open Test File Preset
- Select Regulate
- Select Go on Sweep dialog (single point sweep)
- See Data Editor Table for results

IMD HF (CCIR)

- Open IMD CCIR Standard macro
- Run the macro (macro runs the regulation)
- Open Excel Spreadsheet it creates to see the results

- No graph

Crosstalk

- Open Test File Preset Part 1 (channel 1/L)
- Select Regulate Button
- Select go on the sweep dialog

- See data in data editor table and graph

- Save graph to directory

- Open Test File Preset Part 2 (channel 2/R)
- Select Regulate button
- Select go on the sweep dialog

- See data in data editor table and graph

- Save graph to directory

Common Mode Rejection Ratio

- Open CMRR Test File Preset
- Insert CMRR Cable in channel 1
- Select Regulate
- Choose Common Mode Test (CMTST) on the Analog Generator Dialog
- Highlight the bar graph for the channel that you are planning to test
- While tone is being output, press one 10 Ohm button at a time. Whichever one yields the worse result (the higher (closer to 0) the number) is the one you will test with.
- Holding down that button, Select go on the sweep button

- Read the result from the data editor table and the graph (paying attention to the channel you tested.

- Save graph to directory

- Run again with the other channel, skipping regulation and choosing CMTST.

- Read the result from the data editor table and the graph (paying attention to the channel you tested.

- Save graph to directory

Spurious Aharmonic Signals

- Open test file preset
- Select Regulate
- Select Go on the Sweep Dialog
- See the graph to see if any signal went past the yellow line, ignoring the stimulus at 997Hz and the harmonics. To get the actual reading place the cursor over the biggest spike and read the value.

- Save graph to directory

Amplitude Linearity

- Open Macro
- Play macro
- Open spreadsheet that the test creates and review the standard deviation value in the spreadsheet.

Alias Rejection

- Open test preset file 96kHz
- Select Regulate
- Select Go on Sweep Dialog

- See graph pass/fail line and see log and look for greatest value (giving a ¼ octave leeway from the start of the first frequency before one starts looking at the limit)

- Save graph to directory

Sync Input Jitter Susceptibility

- Open Test Preset file for 12kHz (part 1)
- Open macro for 12kHz (part 1)
- Make sure right samplreate, input and output are correct in the digital IO dialog
- Play macro
- Cursor to the each of the peak signals and measure and/or review the log data and look for the highest value at the specific points.

- Save graph to directory

- Open Test Preset file for 997Hz (part s)
- Open macro for 997Hz (part s)
- Make sure right samplreate, input and output are correct in the digital IO dialog
- Play macro
- Cursor to the each of the peak signals and measure and/or review the log data and look for the highest value at the specific points.

- Save graph to directory

Jitter Transfer Gain

- Open Test Preset File
- Make sure right samplreate, input and output are correct in the digital IO dialog
- Play macro
- Select Go on the sweep dialog
- See graph and log data for results
- Export Graph to directory

- Save graph to directory

End

- Save Log file as, name of org, device, date to directory
- Save spreadsheets to directory

Low cost test procedure

General Comments

Since the goal of the site visits was to capture data and not to report the results on the spot, the consultants focused on capturing the files generated by passing test signals through the device under test and recording the digital output of the device under test to their computer. The consultants then took the resulting files back to New York to perform the analysis and reporting.

Test procedure

Use Y-Cable for testing

Initial Calibration

- Load Recall Config 0 – this is 997Hz at 18dBu
- Change level on Minirator until one sees -1dBFS on the meters of the system being recording to (in this case Samplitude)
- Document the level on the minirator that produces -1dBFS

Frequency Response (20 – 20kHz)

- Load Wave File 1 (This is 1kHz tone at beginning to calibrate to it followed by frequency sweep)
- Set level to -20dBFS
- Select record in Samplitude
- Select play on the minirator file 1
- Record until it goes through the sweep and back to the 1kHz You want to capture 1Khz before and after the sweep in the file.

THD+N

- Load Config 1 (41Hz)
- Set level to level that = -1dBFS (17dBu)
- Record for 10 seconds or so

- Load Config 0 (997Hz)
- Record for 10 seconds or so
- Set level to -1dBFS (17dBu)
- Load Config 0 (997Hz)
- Set level to -10dBFS (8dBu)
- Record for 10 seconds or so
- Change to -20 (-2dBu)
- Record for 10 seconds or so
- Change to -60 (-42dBu)
- Record for 10 seconds or so

- Load Config 2 (6597Hz)

- Set level to -1dBFS (17dBU)
- Record for 10 seconds or so

Dynamic Range

- Use THD+N for 997Hz at -60dBFS
- Load config 0
- Set to -60dBFS (-42dBu)
- Record for 10 seconds or so

Cross Talk

- Load Config 3 (20Hz)
- Set level to -1dBFS (17dBu)
- Short channel 1 using the shorting plug (mimicking output impedance of device)
- Record for 10 seconds or so
- Short channel 2 using the shorting plug (mimicking output impedance of device)
- Record for 10 seconds or so

- Load Config 0 (997Hz)
- Set level to -1dBFS (17dBu)
- Short channel 1 using the shorting plug (mimicking output impedance of device)
- Record for 10 seconds or so
- Short channel 2 using the shorting plug (mimicking output impedance of device)
- Record for 10 seconds or so

- Load Config 4 (20kHz)
- Set level to -1dBFS (17dBu)
- Short channel 1 using the shorting plug (mimicking output impedance of device)
- Record for 10 seconds or so
- Short channel 2 using the shorting plug (mimicking output impedance of device)
- Record for 10 seconds or so

IMD LF (need a peak reading meter. Not an RMS meter for this test to measure the -1dBFS)

- Load Wave File 2
- Set level to -1dBFS
- Record for 10 seconds or so

IMD HF (need a peak reading meter. Not an RMS meter for this test to measure the -1dBFS)

- Load Wave File 3
- Set level to -1dBFS
- Record for 10 seconds or so

Spurious Aharmonic Signals

- Use 997 at -1dBFS recorded earlier for this test

Site Visit Notes

While test results were not tallied and reported at each of the sites, there were circumstances with each visit that revealed insights into the guideline and its application.

National Archives

Chris Lacinak far left, Phillip Sztenderowicz at the bottom center and Ryan Davis of NARA at the top right

The NARA ADC is both an analog-to-digital-converter (ADC) and a digital-to-analog-converter (DAC) containing many input and output channels. However, NARA is using this device as a 2-channel unit, leading us to test the channels under use.

Working with the NARA ADC was challenging. In setting up and getting started with testing the consultants continually saw results that were indicative that a preference or setting was wrong. This led to navigating through the device interfaces to review preferences and configuration settings. What they discovered is that the NARA ADC has an extensive set of features, bells and whistles. There are also multiple user interfaces, consisting of the user interface on the hardware device itself and two different software based interfaces, with very little, if any, overlap. The user interface for preferences and configuration on the hardware device is extremely challenging, with a small LCD display and limited buttons and wheels for controls, it is a maze of menus and submenus that make for a difficult user experience. The multiple user interfaces also make it difficult to know where a given setting might be, or where a given adjustment is best made.

The issues the consultants kept running into were manifestations of this finding, revealing settings that had been inadvertently changed that wouldn't be notable in routine operations. It was only through testing that they were able to identify the issues. A significant amount of time was spent beginning testing, identifying an issue, troubleshooting to resolve the issue and then having to start over again. This raises the peripheral issue of increased risk of operator error that accompanies user interfaces like that of the NARA ADC. These have become increasingly popular with devices that have increasing inputs and outputs and many features and options.

LC Packard Campus at Culpeper, Virginia

From left to right: Robert Friedrich of Library of Congress, Phillip Sztenderowicz, Chris Lacinak and William Haley of Library of Congress

Following the visit to the National Archive the consultants visited the Library of Congress Packard Campus facility for audiovisual preservation at Culpeper, Virginia, where they met with Audio Preservation Specialist Robert Friedrich and Audio Maintenance Technician William Haley. There the consultants tested the LC ADC using XLR analog inputs and AES digital outputs. Testing of this device went smoothly, leaving some time for additional conversation. They knew at the beginning of the project that the Library of Congress Culpeper facility owns and uses an Audio Precision 2700 series test device, and the consultants imagined that one of the deliverables of this project would be templates and scripts that they could provide to users of AP 2700 series devices to use in their routine testing. Discussion confirmed that such deliverables would be welcomed and desirable.

With some of the additional time the consultants were also able to test the stability of another ADC's clock to confirm that it was operating with precision. In other words, there was an interest in confirming that the ADC was indeed operating precisely at the sample rate at which it reported to be operating. For instance, that it was operating at a sample rate of 96kHz when set to 96kHz. They were able to confirm using the AP 2722 that the ADC was in fact well within specification with regard to the clock.

Voice of America

From left to right: Brian Schiff, Jeff Tofani, Russell Mitchell, Chris Lacinak

From left to right: Phillip Sztenderowicz, Jeff Tofani, Russell Mitchell, Chris Lacinak

The following day the consultants visited with several members of VOA, including Russell Mitchell, Telecommunications Manager of Radio Maintenance Service; Jeff Tofani, Project Engineer within Special Projects; and Brian Schiff, Broadcast Engineer. There are many places within VOA where ADCs are employed and where there was interest in utilizing the guideline for testing. Our contacts reserved a room which is utilized for both live recording and digitizing analog sources. The first test was of the VOA high-end ADC.

Loosely related to the issues at the National Archives, the consultants ran into issues in the initial setup with setting levels for the VOA high-end ADC. The card is software controlled, and as is typical within large organizations there were strict permissions on the host computer which disallowed attainment of proper levels. The first image above, showing Brian Schiff kneeling down, is showing Brian troubleshooting the issue. They were never able to fully resolve the issue due to permissions, and it was necessary to perform the tests using different levels.

Another issue the consultants ran into was that the VOA high-end device did not have reference input which kept us from being able to perform the jitter based tests. Therefore there is a notable absence of this data in the test results.

Similar to Culpeper, VOA owns Audio Precision devices but tends to use them in simpler ways to get fast results with high precision. In a broadcast environment they typically have to work with a level of urgency that provides a disincentive to perform more complex testing. They were very interested in obtaining the templates and scripts for use with their devices in order to more easily conduct the testing outlined in the guideline.

VOA was also interested in having us test an ADC that was newly acquired for use by reporters recording stories at their desks or at home. The VOA low-end ADC offered fewer interface options, making it difficult to use the guideline. The consultants spent considerable time attempting to perform the tests with limited success. Ultimately they were unable to test the device, exhibiting the limitations of being able to test lower cost consumer or prosumer devices without the interfaces necessary to perform high performance testing.

Site Visit Results

High Performance

General Information

Date	4/28/15
Organization	National Archives and Records Administration
Time	09:00 ET
Operator	Phillip Sztenderowicz
Analyzer	AP 2722
Serial Number	SYS2-30038
Location	National Archives II
Temperature (F)	78.8
Mains Voltage	113.3

Equipiment Under Test

Manufacturer	- -
Make	- -
Model	NA
Serial Number	- -
Sampling Rate	96000
Bit Depth	24
Notes	Using inputs 1 and 2 via TRS, set to Lo Gain on both channels. Clock source internal for all non-jitter tests

Frequency Response	Smpl Rate	Limit	NARA ADC Left	NARA ADC Right
20 -20KHz	48K	+/- 0.1dB	NA	NA
20 -20KHz	96K	+/- 0.1dB	-0.14	-0.14
20KHz-40KHz	96K	+/- 0.5 dB	-0.03	-0.03

THD +N	Level	Limit (Unw)		
41Hz	-1 dBFS	-100	-104.41	-102.49
997Hz	-1 dBFS	-100	-105.64	-104.63
6597Hz	-1 dBFS	-100	-103.16	-103.03
997Hz	-10 dBFS	-100	-98.29	-98.54
997Hz	-20 dBFS	-90	-92.28	-92.46
997Hz	-60 dBFS	-50	-48.64	-48.27

Dynamic Range (SnR)	Smpl Rate	Limit		
unweighted	48K	-110 dB	NA	NA
	96K	-110 dB	-109.68	-109.6
A-weighted	48K	-112 dB	NA	NA
	96K	-112 dB	Not available	

Crosstalk (interchannel)	Frequency	Limit		
add 1dB to the value in the log file to get final result	20Hz	-110 dB	-94.71	-95.28
The 10kHz test point is just a weigh point because there's a shift in the limit level.	997Hz	-110 dB	-125.33	-130.47
	20KHz	-105 dB	-113.75	-114.00

CMRR	Frequency	Limit		
add 20 and negate the log file result to get final result	60Hz	70 dB	55.18	58.13
	1KHz	70 dB	54.89	57.51
	20KHz	50 dB	54.89	57.47

IMD LF	Limit		
	-100	-98.73	-96.67

IMD HF	Limit		
	-105 dB	-97.75	-97.47

Amplitude Linearity

		Limit		
Reported as Standard Deviation		0.05	0.033	0.046

Spurious Aharmonic Signals

		Limit		
	> 50Hz	-100	-140	-140

Alias Rejection

	Smpl Rate	Limit		
Add 10 to the originally reported result	48kHz	-80	NA	NA
	96kHz	-80	-72	-72

Sync Input Jitter Susceptibility

12kHz	8K	-130dB	Not available
12kHz	4K	-120dB	Not available
12kHz	2K	-120dB	-108
12kHz	1KHz	-120dB	-100
12kHz	500hZ	-100dB	-96
12kHz	250Hz	-90dB	-91
12kHz	125Hz	-70dB	-84
12kHz	63Hz	-60dB	-78
997Hz	500Hz	-110dB	-126
997Hz	250Hz	-100dB	-117
997Hz	125Hz	-90dB	-111.5
997Hz	63Hz	-80dB	-108

Jitter Transfer Gain

	Limit	
	<20ns p-p	601.3 picoseconds

General Information

Date	4/28/15
Organization	Library of Congress
Time	14:30 ET
Operator	Phillip Sztenderowicz
Analyzer	AP 2722
Serial Number	SYS2-30038
Location	NAVCC Audio Room A3
Temperature (F)	71 F
Mains Voltage	121.5

Equipment Under Test

Manufacturer	- -
Make	NA
Model	- -
Serial Number	Unknown
Sampling Rate	96000
Bit Depth	24
Notes	XLR Analog In/ AES In/Out Spectrally flat dither. Clock source internal for all non-jitter tests

			NAVCC ADC	
Frequency Response	**Smpl Rate**	**Limit**	**Left**	**Right**
20 -20KHz	48K	+/- 0.1dB	NA	NA
20 -20KHz	96K	+/- 0.1dB	-0.106	-0.111
20KHz-40KHz	96K	+/- 0.5 dB	-0.39	-0.40

THD +N	**Level**	**Limit (Unw)**		
41Hz	-1 dBFS	-100	-101.36	-98.35
997Hz	-1 dBFS	-100	-100.79	-98.05
6597Hz	-1 dBFS	-100	-100.73	-98.09

997Hz	-10 dBFS	-100	-99.46	-97.63
997Hz	-20 dBFS	-90	-93.24	-92.12
997Hz	-60 dBFS	-50	-64.09	-64.13

Dynamic Range (SnR)	**Smpl Rate**	**Limit**		
unweighted	48K	-110 dB	NA	NA
	96K	-110 dB	-126	-126
A-weighted	48K	-112 dB	NA	NA
	96K	-112 dB	-129	-129

Crosstalk (interchannel)	**Frequency**	**Limit**		
add 1dB to the value in the log file to get final result	20Hz	-110 dB	-146.01	-146.06
The 10kHz test point is just a weigh point because there's a shift in the limit level.	997Hz	-110 dB	-138.69	-133.64
	20KHz	-105 dB	-110.42	-110.06

CMRR	**Frequency**	**Limit**		
add 20 and negate the log file result to get final result	60Hz	70 dB	46.16	45.13
	1KHz	70 dB	46.15	45.14
	20KHz	50 dB	46.28	45.26

IMD LF		**Limit**		
		-100	-90.51	-87.16

IMD HF		**Limit**		
		-105 dB	-103.99	-102.31

Amplitude Linearity		**Limit**		
Reported as Standard Deviation		0.05	0.048	0.107

Spurious Aharmonic Signals		**Limit**		
	> 50Hz	-100	-132.79	-131.94

Alias Rejection	**Smpl Rate**	**Limit**		
Add 10 to the originally reported result	48kHz	-80	NA	NA
	96kHz	-80	-25.14624789	-25.18763193

Sync Input Jitter Susceptibility

12kHz	8K	-130dB	Not available
12kHz	4K	-120dB	Not available
12kHz	2K	-120dB	-94.4
12kHz	1KHz	-120dB	-80.75
12kHz	500hZ	-100dB	-67.5
12kHz	250Hz	-90dB	-58.5
12kHz	125Hz	-70dB	-59.5
12kHz	63Hz	-60dB	-60
997Hz	500Hz	-110dB	-94.3
997Hz	250Hz	-100dB	-85.8
997Hz	125Hz	-90dB	-86.1
997Hz	63Hz	-80dB	-86.8

Jitter Transfer Gain	**Limit**	
	<20ns p-p	15ns

General Information

Date	4/29/15
Organization	Voice of America
Time	09:30 ET
Operator	Phillip Sztenderowicz
Analyzer	AP 2722
Serial Number	SYS2-30038
Location	VOA Studio 2440
Temperature (F)	78 F
Mains Voltage	111

Equipment Under Test

Manufacturer	- -
Make	- -
Model	NA
Serial Number	Unknown
Sampling Rate	48000
Bit Depth	24
Notes	XLR Analog In,set to Lo Gain. Card would only perform at 48kHz so all tests are performed at this sampling rate.

			VOA High-End ADC	
Frequency Response	**Smpl Rate**	**Limit**	**Left**	**Right**
20 -20KHz	48K	+/- 0.1dB	-0.113	-0.104
20 -20KHz	96K	+/- 0.1dB	NA	NA
20KHz-40KHz	96K	+/- 0.5 dB	NA	NA

THD +N	**Level**	**Limit (Unw)**		
41Hz	-1 dBFS	-100	-91.53	-92.28
997Hz	-1 dBFS	-100	-91.50	-92.53
6597Hz	-1 dBFS	-100	-92.32	-92.44

997Hz	-10 dBFS	-100	-84.13	-83.91
997Hz	-20 dBFS	-90	-78.08	-78.09
997Hz	-60 dBFS	-50	-34.18	-34.08

Dynamic Range (SnR)	**Smpl Rate**	**Limit**		
unweighted	48K	-110 dB	-100.21	-100.10
	96K	-110 dB	NA	NA
A-weighted	48K	-112 dB	-103.05	-103.19
	96K	-112 dB	NA	NA

Crosstalk (interchannel)	**Frequency**	**Limit**		
add 1dB to the value in the log file to get final result	20Hz	-110 dB	-101.04	Not Available
The 10kHz test point is just a weigh point because there's a shift in the limit level.	997Hz	-110 dB	-124.33	Not Available
	20KHz	-105 dB	-94.85	Not Available

CMRR	**Frequency**	**Limit**		
add 20 and negate the log file result to get final result	60Hz	70 dB	65.19	65.09
	1KHz	70 dB	64.87	64.28
	20KHz	50 dB	49.19	46.03

IMD LF		**Limit**		
		-100	-80.83	-81.72

IMD HF		**Limit**		
		-105 dB	-98.18	-98.62

Amplitude Linearity		**Limit**		
Reported as Standard Deviation		0.05	0.20	0.12

Spurious Aharmonic Signals		**Limit**		
	> 50Hz	-100	-128.78	-128.92

Alias Rejection	**Sample Rate**	**Limit**		
Add 10 to the originally reported result	48kHz	-80	-84.37	-84.51
	96kHz	-80	NA	NA

Sync Input Jitter Susceptibility

12kHz	8K	-130dB	Not available
12kHz	4K	-120dB	Not available
12kHz	2K	-120dB	Not available
12kHz	1KHz	-120dB	Not available
12kHz	500hZ	-100dB	Not available
12kHz	250Hz	-90dB	Not available
12kHz	125Hz	-70dB	Not available
12kHz	63Hz	-60dB	Not available
997Hz	500Hz	-110dB	Not available
997Hz	250Hz	-100dB	Not available
997Hz	125Hz	-90dB	Not available
997Hz	63Hz	-80dB	Not available

Jitter Transfer Gain	Limit	
	<20ns p-p	Not available

Low Cost

General Information

Date	4/28/15
Organization	National Archives and Records Administration
Time	09:00 ET
Operator	Phillip Sztenderowicz
Analyzer	NTI MR-Pro
Serial Number	ARTA 1.8.4
Location	National Archives II
Temperature (F)	78.8
Mains Voltage	113.3

Equipment Under Test

Manufacturer	- -
Make	- -
Model	NA
Serial Number	- -
Sampling Rate	96000
Bit Depth	24
Notes	TRS Analog In, set to Lo Gain Clock source internal

THD +N	**Level**	**NARA ADC Left**	**NARA ADC Right**
41Hz	-1 dBFS	-87.33	-87.33
997Hz	-1 dBFS	-93.98	-94.42
6597Hz	-1 dBFS	-91.7	-91.7
997Hz	-10 dBFS	-90.75	-90.46
997Hz	-20 dBFS	-82.62	-82.73
997Hz	-60 dBFS	-42.85	-42.97

CROSSTALK (interchannel) **Frequency**

add 1dB to the value in the log file to get final result	20Hz		-112.56	-112.74
	997Hz		-124.91	-128.52
	20KHz		-96.25	-95.91

IMD LF

-82.5	-82.16

IMD HF

-82.38	-82.27

Spurious Aharmonic Signals

-125.44	-122.72

General Information

Date	4/28/15
Organization	Library of Congress
Time	14:30 ET
Operator	Phillip Sztenderowicz
Analyzer	NTI MR-Pro
Serial Number	ARTA 1.8.4
Location	NAVCC Audio Room A3
Temperature (F)	71 F
Mains Voltage	121.5

Equipment Under Test

Manufacturer	- -
Make	NA
Model	- -
Serial Number	Unknown
Sampling Rate	96000
Bit Depth	24
Notes	XLR Analog In Spectrally flat dither Clock source internal

		NAVCC ADC	
		Left	**Right**
THD +N	**Level**		
41Hz	-1 dBFS	-87.13	-86.74
997Hz	-1 dBFS	-94.89	-92.04
6597Hz	-1 dBFS	-92.4	-90.75
997Hz	-10 dBFS	-91.7	-91.37
997Hz	-20 dBFS	-85.68	-85.19
997Hz	-60 dBFS	-49.9	-49.63

CROSSTALK (interchannel)	**Frequency**			
add 1dB to the value in the log file to get final result	20Hz		-157.3	-156.3
	997Hz		-142.94	-142.9
	20KHz		-117.55	-117.9

IMD LF		
	-81.11	-76.48

IMD HF		
	-81.11	-80.82

Spurious Aharmonic Signals		
	-124.58	-125.05

General Information

Date	4/29/15
Organization	Voice of America
Time	09:30 ET
Operator	Phillip Sztenderowicz
Analyzer	NTI MR-Pro
Serial Number	ARTA 1.8.4
Location	VOA Studio 2440
Temperature (F)	78 F
Mains Voltage	111

Equipment Under Test

Manufacturer	- -
Make	- -
Model	NA
Serial Number	Unknown
Sampling Rate	48000
Bit Depth	24
Notes	XLR Analog In,set to Lo Gain. 48 kHz sample rate.

THD +N	**Level**	**VOA High-End ADC Left**	**Right**
41Hz	-1 dBFS	-87.96	-88.18
997Hz	-1 dBFS	-85.51	-85.51
6597Hz	-1 dBFS	-55.39	-55.39
997Hz	-10 dBFS	-80.92	-80.82
997Hz	-20 dBFS	-72.4	-72.04
997Hz	-60 dBFS	-32.54	-32.32

CROSSTALK (interchannel)	**Frequency**			
add 1dB to the value in the log file to get final result	20Hz		-94.69	-95.69
	997Hz		-120.34	-128.34
	20KHz		-127.49	-126.7

IMD LF		
	-81.41	-80.72

IMD HF		
	-92.4	-89.9

Spurious Aharmonic Signals		
	-98.43	-98.23

Challenges

The consultants faced multiple challenges in testing, analysis and reporting for both the high performance and low cost testing.

High Performance

Missing data

Notable in the results is the fact that certain data is marked at “not available”. There were two reasons that contributed to this issue. With the jitter susceptibility tests, the cause of the missing data was traced back to a zoom setting that was changed during our testing at NARA, which was the first of the three organizations the consultants visited. This zoom setting which was changed to demonstrate the test results to those present in the room also impacted the data that was collected as part of the results. They were unaware at the time of this implication and it was only after they returned that they found this to be the case.

The other issue with missing data that came up had to do with the fact that the consultants were continually opening the log file to ensure that all data was being captured. Having the log file open in notepad while performing tests impacted how and where the data was being written, resulting in some of the data being permanently lost.

Both of these issues may be addressed through words of warning in the eventual guideline that is drafted, although they are specific to the AP 2722, and each device and application will have its own eccentricities and behaviors.

Jitter Tests

Upon analysis of the data for jitter susceptibility and jitter transfer gain the consultants realized that the results were so good as to be unbelievable. This led to further investigation that brought about the finding of two issues in the settings of the AP 2722. One led back to the Sync/Ref Input/Output panel within the AP 2722 software. Even though the Digital IO (DIO) page had “Jitter Generation” set to “Sine” the user must still select “Jitter Clock Outputs” in the bottom left hand corner of the “Sync/Ref Input/Output” panel. Figure 1 below shows this panel and option. There were also 2 settings on the “Digital I/O” panel that needed to be changed. One was that the Pk (standing for “peak”) radio button needed to be selected for the measurements they were performing, and the BW (standing for bandwidth) needed to be set to “50Hz to 100kHz”. Figure 2 below shows the relevant panel and options.

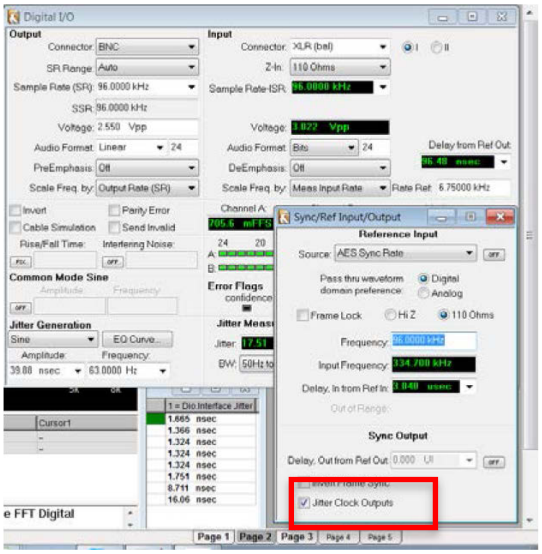


Figure 1

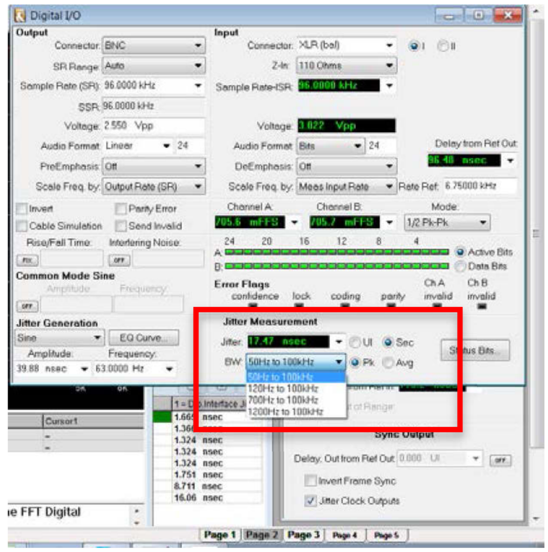


Figure 2

The test results shown above are the incorrect test results and should not be used as evidence of performance for the associated converters. The correct AP settings have now been saved as part of the AP 2722 test files and scripts.

Response

These issues proved the value of “taking the show on the road” which required breaking everything down, setting it back up, and working in a completely different set of environments than the consultants had previously. Aside from being valuable in this way, it also demonstrates just how easy it is to get things wrong. The test instrument itself has hundreds of variables. The devices being tested also have their own set of variables, and the interfaces between the two add further complexity and variables. The ease with which one can make a mistake and generate inaccurate data should be thoroughly considered in thinking through the deployment of these systems and reinforces the original FADGI vision of making the test and measurement process as simple as possible.

In order to address our incomplete and flawed data set the consultants tested additional converters in New York that were used as the basis for making determinations on revisions to the high performance test method and metrics.

Low Cost

The protocol used called for analyzing and reporting on the low cost tests upon returning to New York. The consultants experimented with analyzing the files using analyzers including, the AP 2722, SpectraFoo and ARTA.

The consultants were unable to find a system that could perform the analysis they wanted to perform in non-real time. For the tests employed in the low cost test method, this has little, if anything to do with the technological capability of performing the analysis in non-real-time. In other words, there is no reason that this can’t be done. Moreso, this has to do with the traditional test and measurement workflows and systems which have always been centered on real-time testing. This meant that the consultants had to perform the analysis by reproducing the files in real-time and connecting to an analyzer via AES, essentially “tricking” the system into thinking that it was performing real-time testing.

The system the consultants found to be most promising, and in alignment with the budgetary goals of the low cost setup was ARTA. There was only one test that caused significant challenges and this was the frequency response test. This was due to the fact that the test uses a frequency sweep. Different test and measurement systems will generate frequency sweeps that use different specifications for the frequency range, the total time of the sweep, and the timing of each frequency in the sweep. The analyzer must synchronize to the sweep in order to accurately analyze and report the results. Therefore, a signal generator from one manufacture will not accurately analyze the sweep from a generator that uses specifications that differ from the specification it is programmed to synchronize to. Some analyzers allow the user to program the frequency sweep specifications to enable analysis of various sweeps. Others incorporate automatic detection and synchronization. This feature was not available in the product category that otherwise fit the needs of the low cost test. This feature can be found in expensive devices like the AP 2722, and even in lower cost devices like the NTI Minilyzer. However, the AP 2722 does not fit the budgetary goals and the NTI Minilyzer only offers analog inputs.

Another parameter that proved to be problematic in the low cost test setup was dynamic range. The consultants were unable to test dynamic range in a way that was satisfactory. The test that the ARTA provides consists of testing the ADC channels without any stimulus applied to get a reading of system noise. However, this is not representative of the true dynamic range of ADCs which exhibit significant differences between their static noise floor and the noise floor under stimulated conditions. They believe that it is important to use a measurement methodology that includes a stimulus. The consultants were unable to identify such a methodology available in ARTA. They speculate that one possible method might involve subtracting the THD from the THD+N calculation in ARTA in order to arrive at a noise level that they could potentially use to represent dynamic range. This will require further research, investigation and experimentation.

These findings demonstrated two points. The first is the need for improved test and measurement systems, better suited to the needs of FADGI and the community it serves. The second is the potential need to revisit the non-real-time functionality originally envisioned for the test and measurement protocol. If this latter point proves to be true it will be problematic in fulfilling the goal of a vendor sending files produced as part of their statement of work for the client to analyze as part of their quality control upon receipt of deliverables resulting from a digitization project.

In testing additional converters in New York the consultants decided to circumvent this logistical issue by using ARTA as the signal generator and analyzer, allowing it to perform the frequency response test in real-time. They also took the opportunity to run all other tests utilizing both the MR-Pro and the ARTA combined with the Sound Devices USB Pre2 and a calibration file that ARTA creates to compensate for the nonlinearities of the DAC used as a signal generator source. This allows the opportunity to compare the MRPro signal generator approach to the ARTA signal generator approach.

VI. Guideline Proposals for Two Levels of ADC Performance

This section presents the expert consultants' proposals for two guidelines to be considered by the FADGI Working Group. These proposals are based on the work and outcomes described in sections II through V in this report.

- **Proposed revisions to the 2012 Guideline for High Quality ADC Performance**
- **New proposal for a Guideline for Minimum Quality ADC Performance**

Each draft guideline is followed by an additional statement of the general characteristics of the systems needed to test the performance of ADCs against that guideline.

Normative References (apply to all performance levels)

AES17-1998 (r2009): AES standard method for digital audio engineering — Measurement of digital audio equipment; Revision of AES17-1991. Audio Engineering Society. Retrieved on 2012-08-13 from:
http://www.aes.org/publications/standards/search.cfm?docID=21

Audio Analog-to-Digital Converter Performance Specification and Test Method Introduction. Federal Agencies Digitization Guidelines Initiative. Retrieved on 2012-08-20 from:
http://www.digitizationguidelines.gov/audio-visual/documents/ADC_performIntro_20120820.pdf

IASA TC 04: Guidelines on the Production and Preservation of Digital Audio Objects; Second edition. International Association of Sound and Audiovisual Archives (IASA) Technical Committee. Retrieved 2012-08-13 at:
www.iasa-web.org/tc04/audio-preservation.

IEC-61606-3: Audio and audiovisual equipment - Digital audio parts - Basic measurement methods of audio characteristics - Part 3: Professional use; Edition 1. International Electrotechnical Commission. Retrieved 2012-08-13 at:
http://webstore.iec.ch/webstore/webstore.nsf/artnum/041968!opendocument

IEC 60268-3: Sound system equipment - Part 3: Amplifiers; Edition 3. International Electrotechnical Commission. Retrieved 2012-08-13 at:
http://webstore.iec.ch/webstore/webstore.nsf/artnum/026708!opendocument

IEC 61260-1: Electroacoustics - Octave-band and fractional-octave-band filters - Part 1: Specifications. International Electrotechnical Commission. Retrieved 2015-08-30 at:
https://webstore.iec.ch/publication/5063

Informative References (apply to all performance levels)

Pohlmann, Ken C., *Principles of Digital Audio;* 4th edition. McGraw Hill, 1 - 124

Pohlmann, Ken C., *Measurement and Evaluation of Analog-to-Digital Converters Used in the Long-Term Preservation of Audio Recordings*. Council on Library and Information Resources. Retrieved on 2012-08-13 from:
http://www.clir.org/pubs/resources/ad-converters-pohlmann.pdf

Watkinson, John, *The Art of Digital Audio;* 3rd edition. Focal Press, 1 – 272.

Handbook for Sound Engineers, The New Audio Cyclopedia; 2nd edition. SAMS, 3-42.

Moore, Brian C. J., *An Introduction to the Psychology of Hearing;* 5th edition. Academic Press.

Fielder, Louis D., *Human Auditory Capabilities and their Consequences in Digital Converter Design*, 7th International AES Conference: Audio in Digital Times (May 1989). Audio Engineering Society. Retrieved on 2012-08-13 from: http://www.aes.org/e-lib/browse.cfm?elib=5486

Fielder, Louis D., *Dynamic Range Requirement for Subjective Noise Free Reproduction of Music*, 69th AES Convention (1981). Audio Engineering Society. Retrieved on 2012-08-13 from: http://www.aes.org/e-lib/browse.cfm?elib=11981

Guideline for High Quality ADC Performance: Proposed Adjustments to the 2012 Version

The following tables recap all 12 of the metrics included in the 2012 guideline; highlighting calls attention to the points of adjustment.

<table>
<tr><td>Test Name</td><td>Frequency Response</td></tr>
<tr><td>2012 Test Method</td><td>According to AES-17: Frequency response shall be measured at –20 dBFS with a sinewave whose frequency varies from 10 Hz to 50 kHz in steps no larger than 10 per octave.</td></tr>
<tr><td>2015 Test Method</td><td>Frequency response shall be measured at –20 dBFS with a sinewave whose frequency varies from 10 Hz to 50 kHz in steps no larger than 10 per octave. Results should be reported as a graph and the greatest point of variation shall be documented in dB.</td></tr>
<tr><td>Performance Specification</td><td><table><tr><th>Sample Rate</th><th>Frequency</th><th>Limit</th></tr><tr><td>48kHz</td><td>20 – 20k Hz</td><td>+/- 0.1 dB</td></tr><tr><td>96kHz</td><td>20 – 20k Hz</td><td>+/- 0.1 dB</td></tr><tr><td>96kHz</td><td>20k - 40k Hz</td><td>+/- 0.5 dB</td></tr></table></td></tr>
</table>

Test Name	***Total Harmonic Distortion + Noise (THD+N)***
Test Method	The EUT shall be stimulated with a sine wave. The test signal present in the output shall be removed with a notch filter and bandwidth limited from 20 Hz to 20 kHz. The RMS amplitude is reported as a ratio to the RMS amplitude of the unfiltered signal. The measurement should be performed at the following amplitude and frequency combinations: -1.0 dBFS at 41 Hz, 997 Hz and 6597 Hz, –10 dBFS at 997 Hz, and -20 dBFS at 997 Hz, and -60

	dBFS at 997 Hz.
Performance Specification	<table><tr><th>Freq</th><th>Level</th><th>2012 Limit (unweighted)</th><th>2015 Limit (unweighted)</th></tr><tr><td>Hz</td><td>dBFS</td><td></td><td></td></tr><tr><td>41</td><td>-1</td><td>-100 dB</td><td>-95 dB</td></tr><tr><td>997</td><td>-1</td><td>-100 dB</td><td>-95 dB</td></tr><tr><td>6597</td><td>-1</td><td>-100 dB</td><td>-95 dB</td></tr><tr><td>997</td><td>-10</td><td>-100 dB</td><td>-95 dB</td></tr><tr><td>997</td><td>-20</td><td>-90 dB</td><td>-90 dB</td></tr><tr><td>997</td><td>-60</td><td>-50 dB</td><td>-50 dB</td></tr></table>

Test Name	***Dynamic Range (Signal to Noise)***
Test Method	The measurement is the ratio of the full-scale amplitude to the r.m.s. noise and distortion, expressed in dB, in the presence of signal. It includes all harmonic, inharmonic, and noise components. The test signal shall be a 997 Hz sine wave producing -60 dBFS at the EUT output. Any 997 Hz test signal present in the output is removed by means of a standard notch filter. The results shall be reported as unweighted and A-weighted with a 20 kHz low-pass filter applied, in dBFS. For A-weighted, the remaining noise shall be filtered with an A weighting filter.
Performance Specification	<table><tr><th>Weighting</th><th>Limit</th></tr><tr><td>Unweighted</td><td>-110 dBFS</td></tr><tr><td>A weighted</td><td>-112 dBFS</td></tr></table>

Test Name	***Cross-Talk***
2012 Test Method	One channel of the EUT is driven with a -1 dBFS sinewave and the maximum amplitude of this frequency appearing in any other channel is noted. The measurement is repeated for each input channel and the maximum amplitude for all channels is determined. This amplitude, expressed in dBFS, is increased by 1 dB and reported. The measurement shall be performed at frequencies of 20 Hz, 1 kHz and 20 kHz.
2015 Test Method	One channel of the EUT is driven with a -1 dBFS sinewave. The output of the other channels is passed through a narrow bandpass filter and the maximum amplitude of this frequency appearing in any other channel is noted. The measurement is repeated for each input channel and the maximum amplitude for all channels is determined. The measurement shall be performed at frequencies of 20 Hz, 997 Hz and 20 kHz, and shall be expressed as a ratio, in dB, between the output of the driven channel and the channel under test
Performance	<table><tr><th>Frequency</th><th>Limit</th></tr></table>

Specification	20 Hz	-110 dB
	997 Hz	-110 dB
	20 k Hz	-105 dB

Test Name	***Common-Mode Rejection Ratio (CMRR)***
2012 Test Method	The input shall be driven from a sinewave generator whose output impedance is less than 100 Ohms. The amplitude is adjusted to achieve -20 dBFS at the EUT output. The signal is removed, and the generator reconnected between the chassis ground and one side of the input. A 600 Ohm resistor is connected between this point and the other side of the input. If the input is asymmetrical, the generator should be connected to the low side and the resistor to the high side. The output should be measured through a bandpass filter at the sinewave frequency. The resulting RMS value, measured in dBFS, is increased by 20 dB and reported as a dB (not dBFS) value. The measurement should be performed at 60 Hz, 1 kHz and 20 kHz.
2015 Test Method	The input shall be driven from a sinewave generator whose output impedance is less than 100 Ohms. The amplitude is adjusted to achieve -20 dBFS at the EUT output. The signal generator should then be switched to a common-mode rejection test configuration. Typically this involves the low side signal being directed to the chassis and the high side signal being directed to both the high and low legs routed through well matched resistors (better than .003%). This results in the high and low legs carrying the same signal. Substantial attenuation in the output measurement should be seen in this scenario, as the signal on the two legs should cancel (80 – 90dB of cancellation). For balanced connections, following the output of the signal generator, the insertion of a 10 ohm resistor is alternated between legs and the leg yielding the highest EUT output level is noted. If the input is unbalanced, the resistor should be inserted on the high side. The output shall be measured through a bandpass filter at the stimulus frequency. The resulting RMS value, measured in dBFS, is increased by 20 dB and reported as a positive dB value.

<table>
<tr><td></td><td>The measurement should be performed at 60 Hz, 997 Hz and 20 kHz.

Note that the limit is a lower limit, meaning that passing values are those which are greater than the stated limit.</td></tr>
<tr><td>Performance Specification</td><td><table><tr><th>Frequency</th><th>Limit</th></tr><tr><td>60 Hz</td><td>70 dB</td></tr><tr><td>997 Hz</td><td>70 dB</td></tr><tr><td>20 k Hz</td><td>50 dB</td></tr></table></td></tr>
</table>

2015 Explanatory note re: CMRR
In an effort to clarify this test the following two figures are provided. The first figure is from a paper authored by Bill Whitlock, titled Design of High-Performance Balanced Audio Interfaces, and found at http://sound.westhost.com/articles/balanced-interfaces.pdf. This signal diagram visualizes the CMRR test method language.

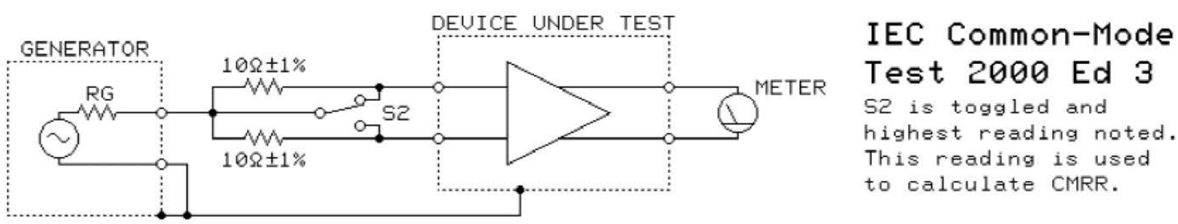


Figure: CMRR test signal diagram

The second figure is a box made by Phillip Sztenderowicz. The 4 momentary-on push-buttons, when pressed, are connected to 600 Ohm and 10 Ohm resistors. This box is placed in between the signal generator and the device under test in order to perform the CMRR test method.

Figure: Box made by Phillip Sztenderowicz to perform CMRR test

Test Name	***Low Frequency Intermodulation Distortion (LF IMD)***
2012 Test Method	Based on AES-17: IM measurements shall be performed with a twin tone signal with a peak amplitude of -1.0 dBFS. The Irms sum of second- and third-order difference

<table>
<tr><td></td><td>frequency components in the output are measured and reported in dBFS.
The test frequencies shall be 41 Hz and 7993 Hz in a 4:1 amplitude ratio.</td></tr>
<tr><td>2015 Test Method</td><td>IM measurements shall be performed with a twin tone signal consisting of 41 Hz and 7993 Hz in a 4:1 amplitude ratio. When summed the signal shall equal -1 dBFS at EUT output. The modulation sidebands below the 7993 Hz tone shall be measured by passing the signal through a 2 kHz high-pass filter and then demodulating, filtering and summing the sidebands. The resulting value shall be reported as a decibel value relative to the amplitude of the 7993 Hz tone.</td></tr>
<tr><td>Performance Specification</td><td><table><tr><th>Frequency</th><th>2012 Limit</th><th>2015 Limit</th></tr><tr><td>LF</td><td>-100 dB</td><td>-90 dB</td></tr></table></td></tr>
</table>

<table>
<tr><td>Test Name</td><td>High Frequency Intermodulation Distortion (HF IMD)</td></tr>
<tr><td>2012 Test Method</td><td>Based on AES-17: IM measurements shall be performed with a twin tone signal with a peak amplitude of -1.0 dBFS. The Irms sum of second- and third-order difference frequency components in the output are measured and reported in dBFS.
The test frequencies shall be 20 kHz and 18 kHz in a 1:1 amplitude ratio.</td></tr>
<tr><td>2015 Test Method</td><td>IM measurements shall be performed with a twin tone signal consisting of 20 kHz and 18 kHz in a 1:1 amplitude ratio. When summed the signal shall equal -1 dBFS. The RMS sum of second- and third-order in-band difference frequency components (ie. 2k, 186, 22k) in the output are measured with a spectrum analyzer or narrow band-pass filter and reported in dB relative to the amplitude of the stimulus.</td></tr>
<tr><td>Performance Specification</td><td><table><tr><th>Frequency</th><th>2012 Limit</th><th>2015 Limit</th></tr><tr><td>HF</td><td>-105 dB</td><td>-100 dB</td></tr></table></td></tr>
</table>

<table>
<tr><td>Test Name</td><td>Amplitude Linearity</td></tr>
<tr><td>2012 Test Method</td><td>Based on AES-17: Level-dependent logarithmic gain is measured at 997 Hz from -5 dBFS to -105 dBFS and reported as standard deviation value in dB.</td></tr>
<tr><td>2015 Test Method</td><td>A 997 Hz sinewave shall be swept from -5 dBFS to -105 dBFS, in steps no larger than 5dB. The amplitude of the output sinewave is measured using a narrow bandpass filter. The deviation in the measured amplitude relative to the the input amplitude is reported as a standard deviation value in dB.</td></tr>
</table>

Performance Specification	<table><tr><th></th><th>Limit</th></tr><tr><td>Standard Deviation</td><td>0.05 dB</td></tr></table>

Test Name	***Spurious Inharmonic Signals (Inharmonic changed from Aharmonic to align with AES-17)***
2012 Test Method	A 997 Hz sinewave shall be applied at -1 dBFS. The output spectrum shall be measued with an 32,768 point FFT. The largest inharmonic component is reported in dBFS.
2015 Test Method	A 997 Hz sinewave shall be applied at -1 dBFS. The output spectrum shall be measured with an 32k point FFT using a Rife-Vincent 5 window. The largest inharmonic component across all channels between 50 Hz and 24 kHz is reported in dBFS.[9]
Performance Specification	<table><tr><th>Frequency</th><th>2012 Limit</th><th>2015 Limit</th></tr><tr><td>50 Hz – 24 kHz</td><td>-100 dBFS</td><td>-130 dBFS</td></tr></table>

Test Name	***Alias Rejection***
2012 Test Method	Based on AES-17 and IEC 61606-3: The device is stimulated with a variable frequency sine wave at -10 dBFS. Beginning at half the sample rate, the frequency is continuously increased until it reaches 200 kHz. For a 48 kHz sample rate, the frequency is swept from 24 kHz to 200 kHz. For a 96 kHz sample rate, the frequency is swept from 48 kHz to 200 kHz. The rms amplitude at the converter output, increased by 10 dB, is graphed. Results are reported as the lowest frequency at which the alias component was equal to or greater in amplitude than all other alias components across the frequency range tested. Amplitude is expressed relative to the stimulus amplitude in dB.
2015 Test Method	The device is stimulated with a variable frequency sine wave at -10 dBFS. Beginning at half the sample rate, the frequency is swept until it reaches 200 kHz. The rms amplitude at the converter output, increased by 10 dB, is graphed. Results are reported as the lowest frequency at which the alias component was equal to or greater in amplitude than all other alias components across the frequency range tested. Amplitude is expressed relative to the stimulus amplitude in dB.
Performance	<table><tr><th>SR</th><th>Limit</th></tr></table>

[9] Application Note: We averaged 8, 32k point FFTs using power averaging and utilized a table sweep to eliminate the harmonic components from being displayed.

Specification	48 kHz -80 dB 96 kHz -80 dB

2015 Explanatory comment re: Alias Rejection

The figure below shows that the calculation is performed by finding the highest aliasing component beyond the initial achievement of alias suppression, and then finding where that matches the sweep that occurs as part of the initial alias suppression. The level is reported as -72 dB and the frequency is reported as 63.1 kHz.

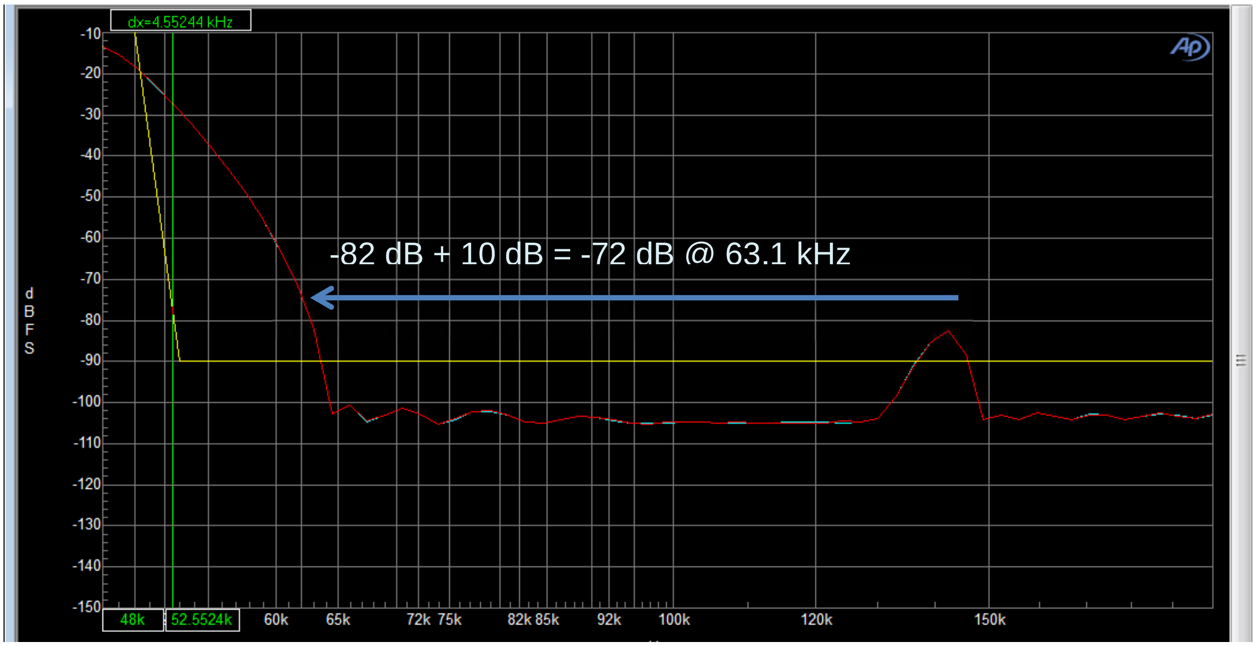


Figure: Explanatory diagram demonstrating measurement of Alias Rejection

Test Name	***Sync Input Jitter Susceptibility***
2012 Test Method	Based on AES-17: The converter input is driven with a -3 dBFS low distortion sinewave at 12 kHz. The reference input is driven with a signal whose phase is jittered with a 40 ns p-p sine-wave whose frequency varies from 62.5 Hz to 8 kHz in octave steps. The output spectrum is measured at each step and the results overlaid. The measurements are repeated with a 997 Hz input to the converter. Results are expressed as dBFS for each octave step.
2015 Test Method	The converter input is driven with a -3 dBFS sinewave at one-fourth the sampling frequency. The clock reference input (not the D/A converter input, if applicable) is driven with a signal whose phase is jittered with a 40 ns p-p sine-wave whose frequency varies from 62.5 Hz to 8 kHz in

	octave steps. The output spectrum is measured using an FFT at each step and the results overlaid. The peak value of each sideband component generated by its associated jitter frequency (i.e. Measured Frequency below) is reported. The measurements are repeated with a 997 Hz input to the converter. Results are expressed as dBFS for each octave step.
Performance Specification	12 kHz

Jitter Frequency	2015 Measured Frequency	2012 Measured Limit	2015 Measured Limit
8 kHz	4 kHz	-130 dBFS	-120 dBFS
4 kHz	8 kHz	-120 dBFS	-110 dBFS
2 kHz	10 kHz	-120 dBFS	-110 dBFS
1 kHz	11 kHz	-120 dBFS	-110 dBFS
500 Hz	11.5 kHz	-100 dBFS	-100 dBFS
250 Hz	11.75 kHz	-90 dBFS	-85 dBFS
125 Hz	11. 875 kHz	-70 dBFS	-70 dBFS
63 Hz	11.937 kHz	-60 dBFS	-60 dBFS

997 Hz

Jitter Frequency	2015 Measured Frequency	Measured Limit
500 Hz	497 Hz	-110 dBFS
250 Hz	747 Hz	-100 dBFS
125 Hz	872 Hz	-90 dBFS
63 Hz	934 Hz	-80 dBFS

Test Name	***Jitter Transfer Gain***
2012 Test Method	Based on AES-17: The reference input shall be driven with a signal whose phase is jittered with a 40 ns p-p sine-wave jitter signal whose frequency varies from 62.5 Hz to 8 kHz in octave steps. The p-p jitter at the output shall be measured at each step and the results shall be graphed. Results shall also report the maximum p-p jitter value in ns.
2015 Test Method	The converter input is driven with a -3 dBFS sinewave at 997 Hz. The clock reference input shall be driven with a signal whose phase is jittered with a 40 ns p-p sine-wave jitter signal whose frequency varies from 62.5 Hz to 8 kHz in octave steps. The p-p jitter at the output shall be measured at each step and the results shall be graphed. Results shall also report the maximum p-p jitter value in ns.
Performance Specification	**Limit** < 20ns p-p

2015 Definitions and Specifications, High-Quality-Level Testing of ADCs

1. Analyzer Specifications

1.1 Standard Notch Filter

The standard notch filter shall have a quality factor *Q* of at least 1.2 and not more than 3, where *Q* is defined as the ratio of the center frequency to the difference between the –3 dB frequencies. Multistage notch filters are acceptable if their combined *Q* measures within these limits using this definition

1.2 Standard Bandpass Filter

The standard band-pass filter shall conform to the class 1 or class 2 response limits described in IEC 61260-1. The attenuation shall be at least 30 dB one octave away from the filter center frequency, and at least 60 dB three octaves away.

> NOTE A filter complying with ANSI S1.11-2004 Class 2 requirements with a bandwidth designator *b* of 2 (that is, a half-octave filter) easily meets this requirement.

If the EUT is very noisy, certain measurements may benefit from the use of a band-pass filter centered on the test frequency to achieve accurate results. Where such measurements are made using a band-pass filter, this shall be noted.

1.3 Narrow Bandpass Filter

A narrow bandpass filter shall have a bandwidth of at least 1/12 octave or a Q of 17.

1.4 THD + N type Distortion Analyzer Specifications

All total harmonic distortion plus noise (THD + N) type distortion analyzers used for measurements in this standard shall utilize a notch filter having an electrical Q of at least 1 and not more than 5. This value shall be verified by measuring the – 3 dB frequencies and computing the ratio of the center frequency to the difference between the – 3 dB frequencies. Multistage notch filters shall be acceptable if their combined Q measures within these limits using this definition. High-pass or band-pass filters should not be part of the measurement path unless specifically required for the test being performed. While such filters may not respond to harmonics only, to be acceptable they must respond to noise, since distortion products which alias in frequency will appear at inharmonic frequencies.

2. Signal Generator Specifications

2.1 Signal Generator Impedance

Unless otherwise specified, the analog signal generators used for measurements in this standard shall have an output impedance of 50 Ohms or less.

2.2 Frequency Accuracy

Signal generators used for measurements in this standard shall provide control over frequency with an accuracy of at least 0,05 %. Alternatively, the frequency may be measured with a frequency counter and adjusted to be within the required accuracy. The frequency adjustment resolution shall be adequate to produce the frequencies specified in the appropriate test.

3. Equipment-Under-Test (EUT) Settings

3.1 General Equipment Settings

The equipment controls shall be set to their normal operating positions except where noted. The switches and controls of the equipment under test (EUT) shall be consistent for all measurements in this standard.

3.2 Emphasis Settings

If any emphasis is provided, it shall be set to the manufacturer's recommended position. This setting shall be clearly indicated in the specifications. If a recommended position is not stated by the manufacturer, emphasis shall not be used. If desired, some measurements may be repeated with other settings, but measurements so obtained shall be clearly indicated as supplementary and shall be reported in addition to the results of the same tests performed using the recommended position.

3.3 Dither Settings

If a dither is provided, it shall be turned on, and this fact shall be clearly indicated in the specifications. If desired, some measurements may be repeated without dither. Measurements so obtained shall be clearly indicated as supplementary and shall be reported in addition to the results of the same tests performed with dither.

3.4 Limiter and Compression Settings

If selectable limiter or compression circuits are included in the EUT, they shall be disabled. If their effect may be measured with additional tests, the results shall be reported separately.

3.5 Device preconditioning

The device shall be connected under normal operating conditions for the manufacturer-specified preconditioning period prior to any measurements being performed. This condition is intended to allow the device to stabilize. If no preconditioning period is specified by the manufacturer, a 5-min period shall be assumed. Should operational requirement preclude preconditioning, the manufacturer shall so state.

3.6 Power interruption

Should power to the device be interrupted during the measurements, sufficient time shall be allowed for restabilization to occur.

3.7 Clock Reference Settings
The clock reference shall be set to internal for all tests with the exception of Jitter Susceptibility and Jitter Transfer Gain

3.8 External Clock Interface
Where external clocking is utilized (i.e. Jitter Susceptibility and Jitter Transfer Gain), the interface used should be an interface dedicated to clock reference interface as opposed to clocking using an interface used for digital-to-analog conversion.

Guideline for Minimum Quality ADC Performance (Partial Metrics): Proposed Performance Guideline

The seven metrics listed in the following table reflect the limitations of the low-cost test system given a field trial in 2012.

Test Name	***Frequency Response***
Test Method	Frequency response shall be measured at –20 dBFS with a sinewave whose frequency varies from 20 Hz to 20 kHz in steps no larger than 10 per octave. Results should be reported as a graph and the greatest point of variation shall be documented in dB.
Performance Specification	**Limit** +/- 0.1 dB

Test Name	***Total Harmonic Distortion + Noise (THD+N)***
Test Method	The EUT shall be stimulated with a sine wave. The test signal present in the output shall be removed with a notch filter and a high pass filter shall be set at 20 Hz. The RMS amplitude is reported as a ratio to the RMS amplitude of the unfiltered signal. The measurement should be performed at the following amplitude and frequency combinations: -1.0 dBFS at 41 Hz, 997 Hz and 6597 Hz, –10 dBFS at 997 Hz, and -20 dBFS at 997 Hz, and -60 dBFS at 997 Hz.
Performance Specification	(see table below)

Freq	Level	Limit (unweighted)
Hz	dBFS	
41	-1	-85 dB
997	-1	-90 dB
6597	-1	-90 dB
997	-10	-85 dB
997	-20	-80 dB
997	-60	-30 dB

Test Name	***Dynamic Range (Signal to Noise)***
Test Method	The test signal shall be a 997 Hz sine wave producing – 60 dBFS at the EUT output. THD is subtracted from THD+N, resulting in a noise value that is expressed in dB.
Performance Specification	**Limit** TBD

Test Name	***Cross-Talk***
Test Method	One channel of the EUT is driven with a -1 dBFS

<table>
<tr><td></td><td>sinewave. The output of the other channel is measured using an FFT. The value measured at the frequency of the stimulus is measured. The measurement is repeated for each input channel and the maximum amplitude across all channels is determined. This amplitude is increased by 1 dB and reported as a ratio in dB. The measurement shall be performed at frequencies of 20 Hz, 997 Hz and 20 kHz.</td></tr>
<tr><td>Performance Specification</td><td><table>
<tr><th>Frequency</th><th>Limit</th></tr>
<tr><td>20 Hz</td><td>-110 dB</td></tr>
<tr><td>997 Hz</td><td>-110 dB</td></tr>
<tr><td>20 k Hz</td><td>-105 dB</td></tr>
</table></td></tr>
</table>

<table>
<tr><td>Test Name</td><td>Low Frequency Intermodulation Distortion (LF IMD)</td></tr>
<tr><td>Test Method</td><td>IM measurements shall be performed with a twin tone signal consisting of 41 Hz and 7993 Hz in a 4:1 amplitude ratio. When summed the signal shall equal -1 dBFS. The amplitudes of the sidebands around 7993 Hz are summed and expressed as dB relative to the amplitude of the 7993 signal.</td></tr>
<tr><td>Performance Specification</td><td><table>
<tr><th>Frequency</th><th>Limit</th></tr>
<tr><td>LF</td><td>-75 dB</td></tr>
</table></td></tr>
</table>

<table>
<tr><td>Test Name</td><td>High Frequency Intermodulation Distortion (HF IMD)</td></tr>
<tr><td>Test Method</td><td>IM measurements shall be performed with a twin tone signal consisting of 20 kHz and 18 kHz in a 1:1 amplitude ratio. When summed the signal shall equal -1 dBFS. The RMS sum of second- and third-order difference frequency components in the output are measured and reported in dB relative to the amplitude of the stimulus.</td></tr>
<tr><td>Performance Specification</td><td><table>
<tr><th>Frequency</th><th>Limit</th></tr>
<tr><td>HF</td><td>-75 dB</td></tr>
</table></td></tr>
</table>

<table>
<tr><td>Test Name</td><td>Spurious Aharmonic Signals</td></tr>
<tr><td>Test Method</td><td>A 997 Hz sinewave shall be applied at -1 dBFS. The output spectrum shall be measured with an 32k point FFT. The largest inharmonic component is reported in dBFS.</td></tr>
<tr><td>Performance Specification</td><td><table>
<tr><th>Frequency</th><th>Limit</th></tr>
<tr><td>> 50Hz</td><td>-120 dBFS</td></tr>
</table></td></tr>
</table>

Definitions and Requirements, Minimum-Quality-Level Testing of ADCs

1. Analyzer Specifications

1.1 Standard Notch Filter

The standard notch filter shall have a quality factor *Q* of at least 1.2 and not more than 3, where *Q* is defined as the ratio of the center frequency to the difference between the –3 dB frequencies. Multistage notch filters are acceptable if their combined *Q* measures within these limits using this definition

1.2 Standard Bandpass Filter

The standard band-pass filter shall conform to the class 1 or class 2 response limits described in IEC 61260-1. The attenuation shall be at least 30 dB one octave away from the filter center frequency, and at least 60 dB three octaves away.

> NOTE A filter complying with ANSI S1.11-2004 Class 2 requirements with a bandwidth designator *b* of 2 (that is, a half-octave filter) easily meets this requirement.

If the EUT is very noisy, certain measurements may benefit from the use of a band-pass filter centered on the test frequency to achieve accurate results. Where such measurements are made using a band-pass filter, this shall be noted.

1.3 Narrow Bandpass Filter

A narrow bandpass filter shall have a bandwidth of at least 1/12 octave or a Q of 17.

2. Equipment-Under-Test (EUT) Settings

2.1 General Equipment Settings

The equipment controls shall be set to their normal operating positions except where noted. The switches and controls of the equipment under test (EUT) shall be consistent for all measurements in this standard.

2.2 Emphasis Settings

If any emphasis is provided, it shall be set to the manufacturer's recommended position. This setting shall be clearly indicated in the specifications. If a recommended position is not stated by the manufacturer, emphasis shall not be used. If desired, some measurements may be repeated with other settings, but measurements so obtained shall be clearly indicated as supplementary and shall be reported in addition to the results of the same tests performed using the recommended position.

2.3 Dither Settings

If a dither is provided, it shall be turned on, and this fact shall be clearly indicated in the specifications. If desired, some measurements may be repeated without dither. Measurements so obtained shall be clearly indicated as supplementary and shall be reported in addition to the results of the same tests performed with dither.

2.4 Limiter and Compression Settings

If selectable limiter or compression circuits are included in the EUT, they shall be disabled. If their effect may be measured with additional tests, the results shall be reported separately.

2.5 Device preconditioning

The device shall be connected under normal operating conditions for the manufacturer-specified preconditioning period prior to any measurements being performed. This condition is intended to allow the device to stabilize. If no preconditioning period is specified by the manufacturer, a 5-min period shall be assumed. Should operational requirement preclude preconditioning, the manufacturer shall so state.

2.6 Power interruption

Should power to the device be interrupted during the measurements, sufficient time shall be allowed for restabilization to occur.

VII. Conclusion and Recommendations for Future Actions

General Findings

1. Result: progress made in 2015

The field tests carried out in three federal agencies in April 2015 provided an important focal point for the year's activity and, as planned, served as a proving ground for the technologies and methods under development. The months preceding the field tests drove the preparation of the systems to be tested, and the descriptions in sections II, III, and IV spell out some of the challenges and difficulties associated with that preparation. The test itself revealed some additional challenges and these are documented in section VI.

The months following the field test included analysis of the challenges and the responses to them, followed by the drafting of this report. The analytic work also saw the development of the conceptual model outlined in section I of this report.

Regarding the high metrics, high cost test system, 2015 saw a pair of tangible outcomes. First, as described in section II, the details for the test system were settled at a reasonable level, although some additional refinements will be featured in work planned for 2016. Second, as presented in section VI, the activity yielded a proposal for adjustments to the 2012 FADGI high quality guideline.

Regarding the minimum metrics, low-cost system, 2015 also saw two tangible outcomes. First, as described in sections III and IV, the project team developed an initial instance of a low cost system capable of carrying out a partial test for minimum metrics. Second, as presented in section VI, the team drafted a proposed partial guideline for minimum ADC performance, featuring elements that fit the capabilities of the low-cost system.

The overall FADGI ADC-testing project continues to represent a strong albeit informal synergy with three other audio preservation efforts. First, it provides a partial response to recommendation 2.4 of the *National Recording Preservation Plan*, "Preservation Workflows for Audio Materials."[10] Second, the detailed technological findings and recommendations described in this report complement the *ARSC Guide to Audio Preservation*, drafted under the auspices of the National Recording Preservation Board.[11] Third, the principle author of this report (Chris Lacinak) has maintained his active and ongoing communication with the Audio Engineering Society standards committees, with special connections to the new AES Project AES-X217.[12]

---

[10] http://www.loc.gov/programs/static/national-recording-preservation-plan/publications-and-reports/documents/NRPPLANCLIRpdfpub156.pdf
[11] http://www.clir.org/pubs/reports/pub164
[12] http://www.aes.org/standards/meetings/init-projects/aes-x217-init.cfm

2. Finding: comprehensive testing of ADCs at high quality levels requires expensive support equipment and engineering expertise

The 2015 execution of comprehensive tests verifies that in order to test an ADC against all 12 metrics in the 2012 guideline requires (a) an expensive audio analyzer and (b) an operator with good engineering skills. Elements like automation-support software can make the task easier, and the development of such software is tentatively planned for 2016. Nevertheless, the need for both an expensive analyzer and an operator with good engineering skills motivated the Working Group to push ahead with the development of a minimum metrics, low cost system, like the one described below.

The comprehensive high metrics system tested in 2015 employed an Audio Precision SYS-2722 Analyzer. This is now a discontinued model. Nevertheless, since instances of this analyzer are owned by a number of archives, including two FADGI members, and since others in the preservation community may have an interest in this topic, the Working Group is providing the scripts and SYS-2722-specific test routines as an appendix to this document on the FADGI website.[13]

3. Finding: the lowest cost systems cannot test for the highest level performance, even if only some of the metrics are tested

During 2015, one low cost system was brought to a proof-of-concept level, and field-tested in federal agencies. The setup and field-testing demonstrated that the equipment and software employed in this system provided some valuable assessment of ADC performance. However, this system was not capable of the accurate and precise measurements at a moderate-to-high performance level. In addition, this type of equipment and software is not capable of measuring all 12 metrics listed in the 2012 guideline.

As noted in the conceptual framework section in the introduction to this report, the 2015 project brought to light the differences in the ways that the high cost and low cost measurement tools execute the measurements. The low cost system's ability to measure several performance features does not provide a simple "lower performance number" when compared to the reading from the high cost system. It is also the case that the low cost system itself has limited capabilities. The ADC being tested might perform better than what the low cost system reports, due to the testing system's limitations. For this reason, the system was dubbed *minimum metrics, low cost*. As noted in the next section of this conclusion, the project will next look at a system with elements that are a bit more expensive (but still low cost, i.e., under $2,500) and that can measure *moderate metrics*.

The consultants emphasize, however, that even the minimum metrics, low cost test system will succeed in identifying genuine failures in ADC operation, even if it is a high performance unit.

---

[13] The scripts for the analyzer are provided as is, with no warranty, to be used by downloaders at their own risk.

Regarding the minimum metrics, the 2015 activities found that this type of system is capable of testing 7 metrics at a minimum performance level. The field test suggested that this system would require an operator with a reasonable grasp of audio engineering, a finding that needs to be confirmed by future use-testing.

4. Finding: the 2015 outcomes with a low cost system indicate the value of making a second try with a more elaborate, moderate cost system

The value of developing a second low cost system emerged as the 2015 project proceeded. The Working Group and the expert consultants developed a conceptual framework for the overall effort and this helped clarify the value of developing a second low-cost (or moderate-cost) system. Dubbed the *moderate metrics, low cost* system, this system would be capable of evaluating performance at a moderate-to-high level for something like 8 or 9 metrics. The exploration of this system will continue in 2016.

Recommendations for system improvements

Throughout 2015, while the activities described in this report were being carried out, the expert consultants had a number of insights regarding opportunities for improving upon cost, quality, and ease of performing ADC testing and the development of tools to do that testing. The following sections list the more salient, notable and pragmatic of these opportunities. The Working Group hopes that some of these will be worked out during 2016; others will await future activities.

1. Use analog attenuation in concert with digital to analog converter to produce a higher quality test signal source.
This opportunity relates to using a DAC as the output of a software based signal generator. The issue that arises with using DACs in this role tends to be their own limited performance. However, there are possibilities for greatly improving upon their performance. This begins with playing test signals at a level close to full scale in order to achieve optimal DAC performance. Using digital attenuation would result in poorer signal to noise performance. This can be overcome by placing a simple, high quality analog attenuator at the output of the DAC, avoiding the need to attenuate digitally. Using analog attenuation, when the signal is decreased, both the signal and the noise are decreased proportionally so performance is not worsened. Such an attenuator could be built relatively cheaply, and even designed using an open-hardware approach, allowing others to build their own, or have it built for them. Furthermore, an analog low pass or band pass filter can be applied to the output in order to decrease the noise floor to an even greater extent, in addition to removing any harmonic content created as an artifact of the DAC. In other words, improving both the signal to noise ratio as well as the overall THD+N performance.

2. Use calibration files to correct for any minor non-linearity in test source.
This opportunity is likely limited to low cost testing, although it may have extended application. The potential here was realized through use of the ARTA software application and one of the functions it offers. The ARTA has the ability to incorporate a calibration file based on the inverse of the calibration file as played back by the DAC in

order to correct for the DAC non-linearities. Initial testing seems to indicate that testing using a DAC with this calibration file approximates the result patterns and trends of the AP 2722. If this method is found to be sufficient it could eliminate the troubling problem of having to use a DAC in order to use ARTA. Using a DAC is troubling because of the variability of DACs that will be used across users, and the inconsistencies in performance that each of them will introduce. Without some way of greatly mitigating the DAC as a variable, this approach would not work well as a uniform standard. This is the reason that the MR-Pro was originally selected. However the challenges faced in using the MR-Pro make use of ARTA with a DAC attractive. For frequency response in particular the calibration file eliminates the inherent non-linearities found in DACs. It must also be stated that using a DAC, even with this method, still currently limits the performance on some tests. The images and text below demonstrate this particular functionality in ARTA.

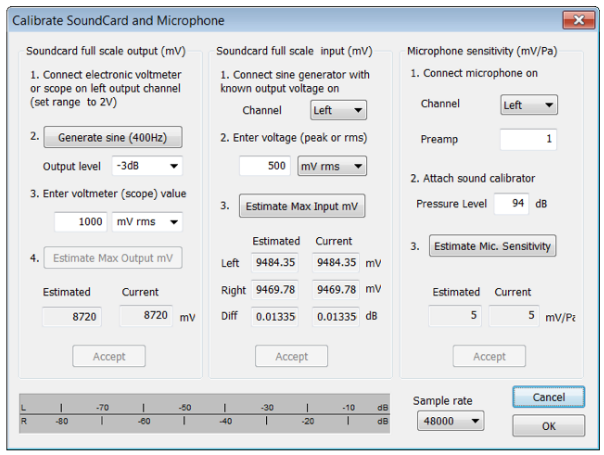
Under Setup>calibrate devices you will find the panel that allows one to calibrate the inputs and outputs of Steps, and Arta.

In this case, we are only interested in the output calibration since the measurements of the input signal will be digital which are inherantly calibrated.

The panel walks you through the process. You will need a voltmeter with true RMS reading capability for this step.

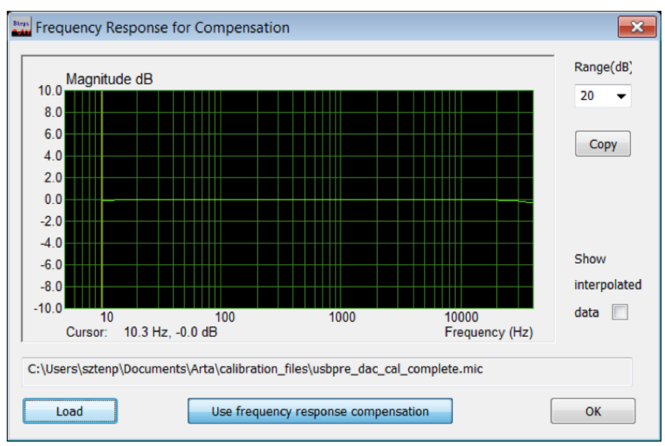
Next, you will need to meaure the frequency response of the audio device D-to-A in order to generate a calibration file to eliminate its non-linearity from the measured response of the device under test. For information on how to do this, refer to the user manual

The calibration file is a simple ascii text file, renamed with a xxx.mic extension (instead of xxx.txt). To instantiate it into the measurement process, select the menu under Setup> Frequency compensation, and then hit the load button to navigate to where your calibration file is stored. Once it's successfully loaded, you will see the compensation curve in the graph. Engage the frequency compensation curve by hitting the' use freqency response compensation' button.



Example Frequency Response Compensation File.

X (Hz)	Y (dBu)
10.25	-0.04765086
10.99	-0.04715086
12.45	-0.04605086
13.92	-0.03585569
15.43	-0.02924782
16.86	-0.02505063
18.33	-0.02217297
20.54	-0.02031567
22.01	-0.01958460
23.53	-0.01924477
24.93	-0.01491483
....	
37512.58	-0.28082721
39743.51	-0.30780740
42107	-0.34580669
44611.1	-0.41445014
47263.89	-0.50966816
50074.16	-0.60033079



3. Confer with manufacturers to modify their techniques to align with the FADGI Guideline.

For both the high performance and low cost guidelines we found multiple scenarios where a given manufacturer’s technique for signal generation and/or measurement did not align with the FADGI guideline test method. Most of these cases were not technical limitations, but rather simple differences in approach. It could be a worthwhile venture to reach out to manufacturers of systems to hold exploratory conversations about aligning their products with the FADGI guideline. Past FADGI surveys on the topic of ADC test and measurement could be used to bolster and support these conversations. Perhaps a one or two day summit, bringing test and measurement experts to the table to discuss this topic could generate momentum that would be advantageous to the advancement and adoption of the FADGI guidelines.

4. Make CMRR measurement tools for standard signal sources more readily available

The CMRR test method appears to most readers to be complex and confusing. In practice it is quite simple. However, this simplicity is primarily based on having two specialized peripherals available. One is a specialized cable/switch in order to momentarily place a 10 Ohm resistor in line, first for assessing how to run the test, and then for running the test. This cable/switch can be seen in the image below. This unit was built by collaborator Phillip Sztenderowicz, an expert who was able to create this cheaply and quickly.

This particular cable is balanced XLR on both ends and allows inserting both 10 Ohm and 600 Ohm resistors in line. However, moving forward only 10 Ohm capabilities are required. To make is as easy as possible for users it would be ideal to have balanced, unbalanced, stereo, and mono versions of this peripheral for sale. In addition to this an open hardware schematic could be drawn and published so that others could make their own or have someone else make it for them.

The other aspect of the CMRR test that not all signal generators provide an option for is running in a common mode configuration, sending the high leg signal to both the high and the low leg, while sending the low leg signal to the chassis. Creating a box that could be inserted at the output of a generator to perform this function, and having it readily available would make this test much more accessible to regular users. Otherwise we fear that many people will simply not perform this test.

5. Encourage the development of tools and methodology to augment the low cost system that would allow amplitude linearity test signals to be generated
Amplitude linearity requires a 1 kHz source to be swept from -105 dBFS to -5 dBFS. This is difficult for most lower cost signal generators to do with precision and with quality. It is feasible that this issue could be tackled by creating this test into two or more parts and associated signal sweeps. For instance, from -5 to -55 dB, and then from -55 to -105 dB. This would minimize the non-linearity of the source, allowing for more accurate testing.

6. Establish tools and procedures to permit Alias Rejection testing of the ADC low pass filters.
Alias rejection continues to prove its worth as a test. However it is not able to be performed with the low cost test because of the limitations of lower cost signal generators. The test requires a signal up to 200 kHz while most lower cost signal generators are 48 kHz or 96 kHz and are limited to half of their sampling frequency. It is feasible, using kits like those offered by Digilent that a board could be produced relatively cheaply that could produce a test signal meeting the specification, which would enable the performance of this test for the lower cost test setups.

7. Encourage the development of tools to facilitate jitter signal generation to permit testing of ADC rejection of external clock noise.
The lower cost tests are unable to perform the jitter based tests because they lack a clock source reference signal that can be used as defined in the test method. It is possible that a standard analog generator could be used as the source to jitter a digital signal. This would require building a box that sums a clock source and the input from the analog generator. This summed signal would then be fed to the clock reference input of the device under test. The clock source could either be derived from a DAC by taking its digital output and using a PLL receiver chip in order to derive the clock from it. It is anticipated that the research and development to develop this would be expensive but the hardware could ultimately be made at a reasonable cost once the research and development was completed.