TECHNOTE 010

AK4619 ADC AND DAC PERFORMANCE

EVALUATION OF THE IMPACTOF A²B I²S CLOCK JITTER

Rev 2

22-May-23

Signal Processing

http://clk.works/

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Clockworks' TechNote008 investigated the jitter properties of the I^2S clock output of the AD2428 A^2B transceiver part. TechNote009[1] made measurements of the impact of A^2B generated clocks on the performance of an AKM AK4490 DAC. From that analysis we saw that designers of mid-range and above of consumer and pro audio applications, as well as related industrial, scientific, and sensing applications that need highly accurate ADC and DAC performance must evaluate the planned hardware platforms to ensure performance is not unexpectedly degraded.

The AD242x A^2B transceiver device's output clocks have high jitter in comparison to I^2S clocks derived from crystal oscillators that are typical in ADC/DAC systems. The performance effects of the A^2B transceiver clock output jitter might not be of concern if the chosen ADC and/or DAC had documented jitter reduction performance and it was possible to pick parts based on those datasheet numbers. Real life is much crueler, with nothing more than a passing marketing reference to "great jitter tolerance" but little to no published data to define what that means for parts examined to date.

This TechNote looks at the AKM AK4619 Quad ADC/DAC, which is a mid-performance part that finds use in audio applications needing higher channel counts. The datasheet does not make any claims about the part's performance when the inputs clock are non-ideal with respect to jitter.

If you want to skip the background and jump to the results, see **Chapter 4 A2B** system + EVM Measurements.

1 INTRODUCTION

A test environment[2] based around AKM's AKD4619 EVM platform was used with Clockwork's standard A^2B hardware products. ADI's SigmaStudio tool was used to configure the different test configurations.

[1] TechNote009 was used as the basis for this document so there's some level of replication in the background information.

[2] This environment is similar to that used for the earlier A^2B studies. One change is the use of Spectral Measurements dScope M1 as the test signal generator and analyzer.

Figure 1 shows the SigmaStudio schematic for the full 10 node configuration. The root (A²B master) node is ADI's WDZ eval board; it includes Toslink input and output so is convenient for providing digital audio to/from the dScope M1 audio analyzer.

Two Clockworks AB0109 Quad module carrier boards provide a compact way to increase node count. The last node is Clockworks' AB0106L which provides I²S input and output that can be directly connected to the AKM EVM's I²S header.

When testing with less nodes one or both quad carrier boards are removed from the daisy chain. The AKM EVM was located at the first location (client node 0 in A²B numbering convention), the 5th location (node 4), and the 9th location (node 8).

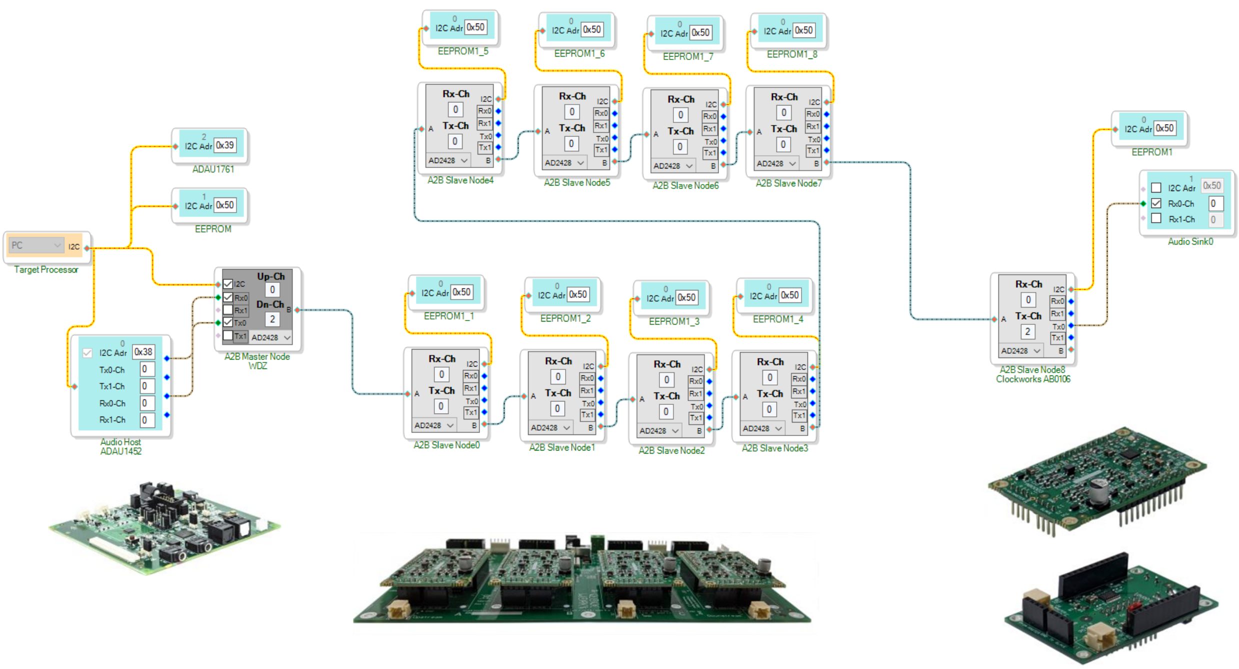

Figure 1 10 node configuration used for test (other configurations are subsets of this)

Clock jitter specifications for the AD2428 A²B transceiver are of possible concern because of this table in the datasheet:

Table 10. SYNC Output RMS TIE Jitter at Each Slave

Slave Node	Typ	Max	Unit
1	1.57		ns
2	1.79		ns
3	1.91		ns
4	2.04		ns
5	2.15		ns
6	2.27		ns
7	2.44		ns
8	2.47		ns
9	2.58		ns
10	2.70	5.50	ns

Figure 2 RMS TIE table from AD242x datasheet

t_{SOJ} from the parameters in Table 7 of the AD2428 datasheet specify 2.2 nsec typical and is assumed to be an average derived from the above. Regardless, this is an unusually high number for audio designers where typical audio band jitter from crystal-based clocks may be a few dozens of picoseconds at most.

Clockworks performed jitter measurements on the A^{2}B hardware and published them in TechNote008 A^{2}B I^{2}S Clock Jitter. For those not familiar with the topic of jitter and audio please see Clockworks Appnote001 – Jitter Spectrum measurement with a DSO.

1.1 SYSTEM PERFORMANCE IMPACT

Complicating matters is that ADCs and DACs all respond to clock jitter differently, and theoretical analysis is difficult and only even possible if the detailed design of a part was available for inspection. With pro audio aiming for Dynamic Range (DR) > 120 dB (AES-17 measurement) and high-performance consumer systems having DR > 110 dB, system hardware measurement and test is the only definitive way to evaluate the impact of the A^{2}B jitter.

For sigma-delta converter architectures, sample clock jitter can manifest itself as an increase in the levels of harmonics, an increase in the noise floor level, or both.

To assess A^{2}B related jitter on the AK4619 the measurements focus on noise, DR, and THD+N measurements. There are other types of tests that may be desirable in evaluating ADC or DAC performance, for example various IMD and TIM (or more generically time domain performance) type tests, but for the focus on clock jitter effects those other performance metrics will not be considered. With current device technology THD+N has been observed to have reasonably good correlation with other the other measurements.

As it turns out, only the THD+N performance of the AK4619 is impacted by the A^2B related jitter.

1.2 A^2B SPREAD SPECTRUM MODES

To pass EMI tests most modern system include a spread spectrum mode that intentionally jitters a clock to cause the emissions to be spread across a wider bandwidth,[3] and A^2B is no exception. Both the A^2B bus and the I^2S ports can be jittered under software control.

The tests here are all performed with the spread spectrum modes disabled as they theoretically only make matters worse.

In a real system it is possible the degradation is small and passing EMI certification will be easier. If you're making a high-performance audio system the approach should always be to fix the EMI problem first as spreading noise doesn't reduce the total noise power.

2 TEST CONFIGURATION

- Root (A^2B Master) node: Analog Devices EVAL-AD2428WD1BZ (controlled via USBi from PC with SigmaStudio)
- Intermediate client (A^2B slave) nodes: Clockworks AB0001 modules in AB0109 carrier.
- Last node: : Clockworks AB0003 module with AB0106L I/O expander.
- AKM AKD4619 EVM
- dScope M-1 Audio Measurement Platform by Spectral Measurement.
  - FFT window set to Prism-6.
  - Unless noted otherwise 64K point FFT used, with 8 averages.
  - Test tone is 997 Hz.

Power for the EVM is from a 5V switching wall wart. This was done to represent the typical noisy supply that local linear regulators might need to deal with. The AKM EVM uses LT1963 low noise[4]

[3] As opposed to reducing the cause of the emissions. Spreading EMI creates problems for most modern wireless systems which use "wide" channels. Though if the noise modulated down into audio frequencies things that raise the noise floor are less objectionable than little whistling birdies.

[4] 40 uVRMS 10 Hz to 100 kHz

LDOs. In the test setup one 5V supply is used to power all sections of the EVM. The EVM was set up with all ground jumpers in place and a common ground for the board.

A laptop was used for the EVM's USB control. The M1's USB port was connected to a desktop PC. The test location has been previously observed to have noticeable 60Hz and related harmonic present as well as there's close proximity (about 1.4 km) to a large FM station's transmitter/tower. Mitigation was attempted but some residuals show up in the results.

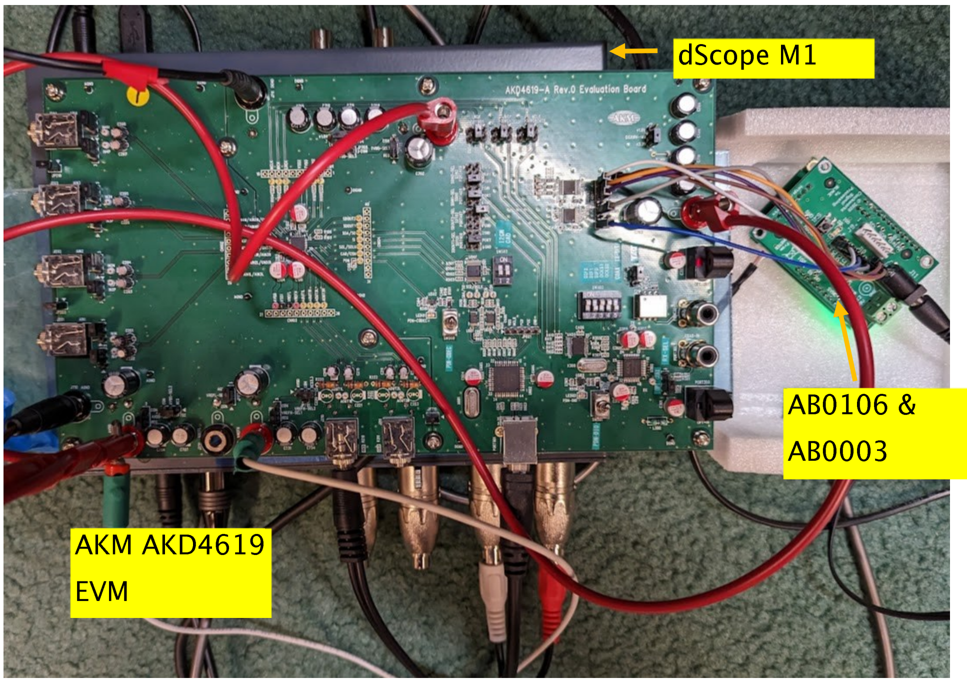

Figure 3 AK4619 testing with M1 and A²B interface (shown connected for DAC testing). See Figure 4 for the rest of the A²B hardware.

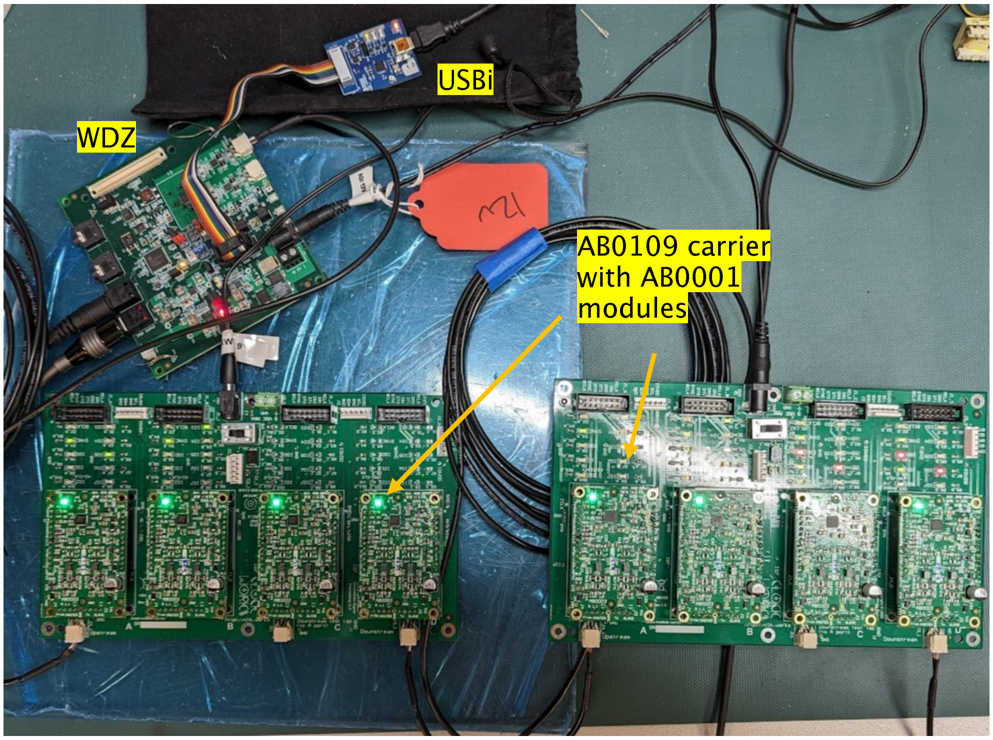

Figure 4 Test hardware setup for testing with A2B, root nodes plus 8 additional nodes used to place AK4619 EVM a large number of hops away from the root to increase clock jitter.

2.1 TEST MEASUREMENT LIMITS

Modern audio equipment can achieve "straight wire with gain" in terms of reducing audible artifacts an order of magnitude or more below even the most optimistic hearing thresholds. With the capability of wide dynamic range and low distortion has come the need for even more capable test equipment. With such sensitive equipment minor outside perturbations can dominate the measurements.

Before performing measurements a check of the test equipment is done to ensure the validity of the measurements.

The dScope M1 system numbers that are relevant to the analog measurement here are specified for a 22 Hz to 22 kHz bandwidth with no weighting:

- THD+N -108 dB + 0.6 uV (note there is auto ranging so the THD+N is not constant at higher signal levels)
- Residual input noise < 0.8 uV (-121 dBV)

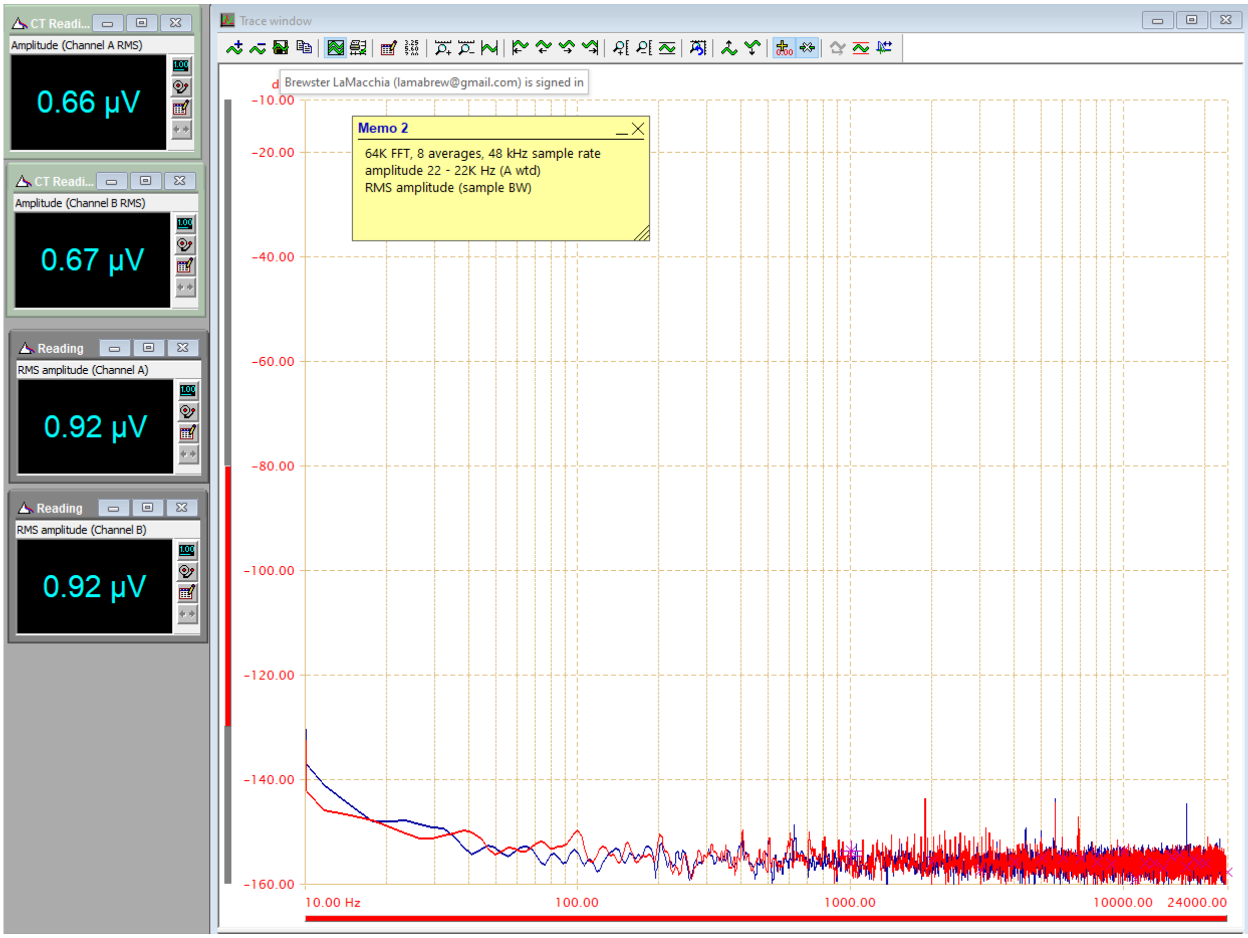

Figure 5 M1 analyzer with input shorted (with RCA cable)

With the cable and other possible external noise from this setup the observed (un-weighted) noise is 120 nV larger (1.2 dB) than the datasheet value. The small (< 100 nV) noise tones' source is not known and attempts to isolate the source(s) have not yielded useable results. As detailed in 6Appendix A, the nature of the noise is not fixed.

Figure 6 is a loopback test with no signal. Residual noise for the output is specified as 1.25 uV (max). With measured input noise of 0.92 uV we could assume the output noise is around 0.88 uV to produce the measured 1.27 uV in Channel A. Some 60 Hz pickup is occurring, the physical test location is bothered by enough 60 Hz background as well as a nearby high-power FM transmitter that getting a perfectly flat noise floor is not possible. Ground loop problems from the PC being used have also been shown to be a factor.

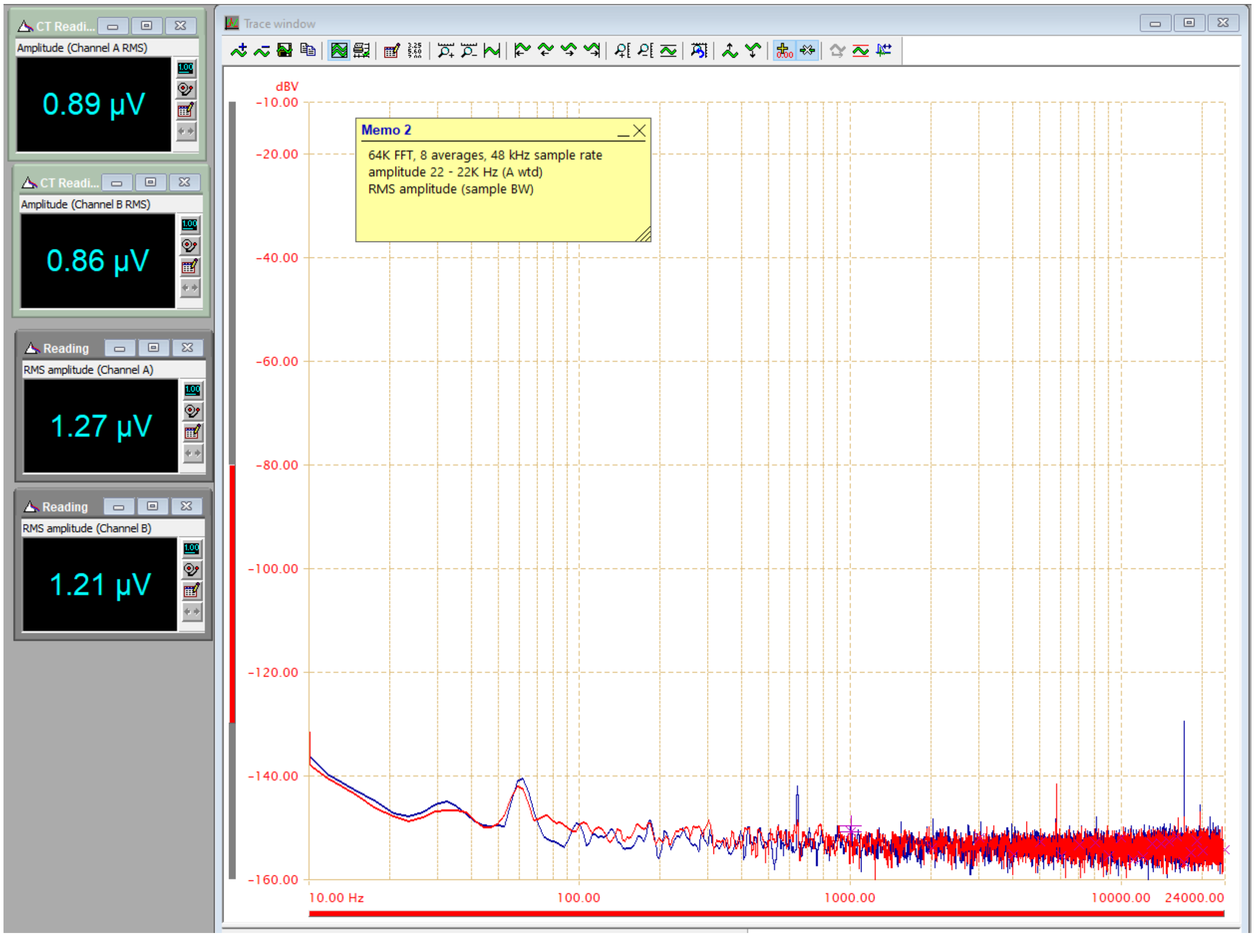

Figure 6 Output to input loopback with RCA cable and no signal.

2.1.1 LOCAL MEASUREMENT LIMITS

Lots of things can conspire to reduce theoretical performance. The dScope M1's output was set to a 1V (0 dBV) level to correspond to the AK4619's full scale values.

At 0 dBV level the measured THD+N is −105 dB. The generator and analyzer are both spec'd at –108 dB plus some noise, so this would indicate the actual system performance is close to the numbers on the datasheet. The harmonic components total to −109 dB as shown in Figure 7.

By comparing measurements with the specific harmonics here i.e., −111 dB for 3rd, −118 dB for the 5th, and −137 dB for the 7th, we can make some observations about the EVM's distortion performance beyond the THD+N number. We're following the common practice of not worrying much about even harmonics that are at low levels and focusing on the odd harmonics that show the effects of circuit non-linearities most readily in a system.

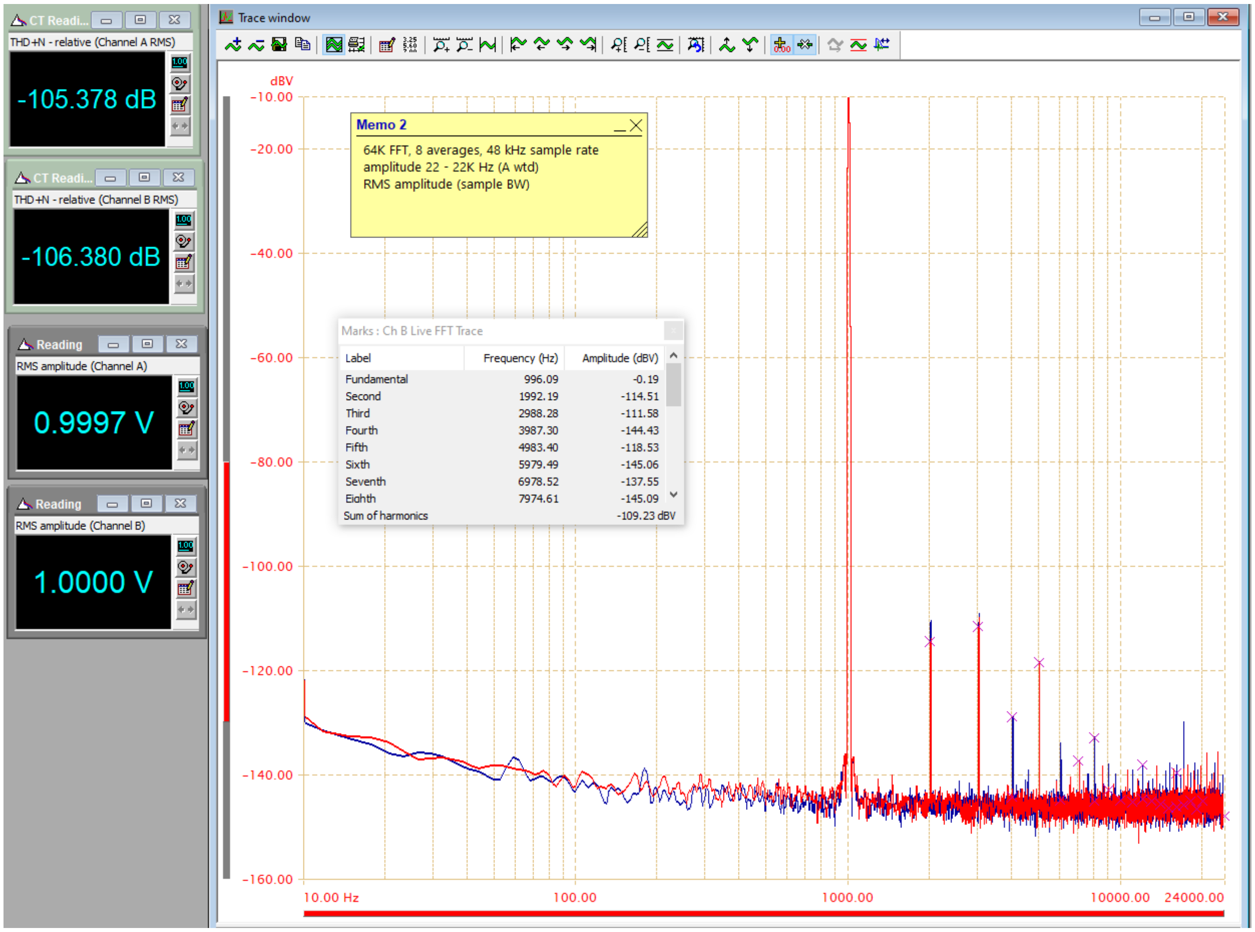

Figure 7 Measured harmonics for dScope M-1 loopback for channel B.

Figure 8 is a THD+N[5] vs. level test run with loopback to show the effects of the analyzer's auto-ranging. When the system switches to a lower gain to accommodate the larger signals it makes the relative THD+N look higher as the noise floor appears to shift. It can also be observed that there is a few dB difference between the two channels.

[5] Unless noted otherwise THD+N tests use a frequency of 997 Hz as it requires the ADC or DAC to use more PCM codes over the measurement window than 1 kHz does. This should produce a better measurement, though ideally delta-sigma architectures shouldn't have issues like that seen with ladder-style converters.

Figure 8 THD+N vs. level to show effect of auto ranging, Brown = Chan A, Purple = Chan B

There is test equipment with better THD+N performance available, but it doesn't help us learn anything practical about the specific ADC/DAC being tested here.

THD numbers of -60 dB can not be heard by most people on actual musical content and even the most sensitive of ears on contrived tests have shown no ability to detect typical harmonics at the -80 dB level.[6] With measurement instrument performance a solid 20 dB better than these limits it becomes an academic exercise to chase higher measurement resolution.

[6] THD+N is not always the best metric to consider what might be audible as there are both frequency and level dependencies; musical content is also quite complex compared to test tones.

Noise levels may be of concern for high performance audio systems set up in an environment that can take advantage of a wide dynamic range.

1. At -60 dBV the measured loopback THD+N is -66 dB. This could be interpreted as 126 dB DR, though the auto ranging of the M1 unit makes that not be a good way to think about it.
2. With shorted input the measured noise floor is -118 dBV unweighted and -123 dBV weighted. This is almost an order of magnitude better sensitivity than what the in-system performance of the AK4619 is.

2.2 TRUST, BUT VERIFY

Before starting a measurement run, and sometimes even during, it's a good idea to recheck that the measurement system is performing as expected. There are an awful lot of things that can go wrong when your measuring instrument can see < 1 uV signals riding in an audio signal a million or more times larger. The specifics here are not meant to be a critique of the M1 unit making these measurements, but as an illustration of the subtle things that can happen.

2.2.1 REVERIFICATION OF THE M1 ANALOG INPUT NOISE MEASUREMENTS

Figure 9 shows the M1's input shorted noise floor and it can be observed that there's a little bit of hair in the spectrum that would not be expected in an ideal environment. Noise floor checks on other days produced different results, for more see (Appendix) Section 6A.1.

There's also a significant difference in asking "what is audible", "what is objectionable", and "what do I think sounds better." Transducers (speakers, etc.) may have distortion of 1% or greater and nobody finds it objectionable. The limit numbers provided here are the author's opinion after considering dozens of journal articles on this topic over several years. The rabbit hole on THD is wide and deep.

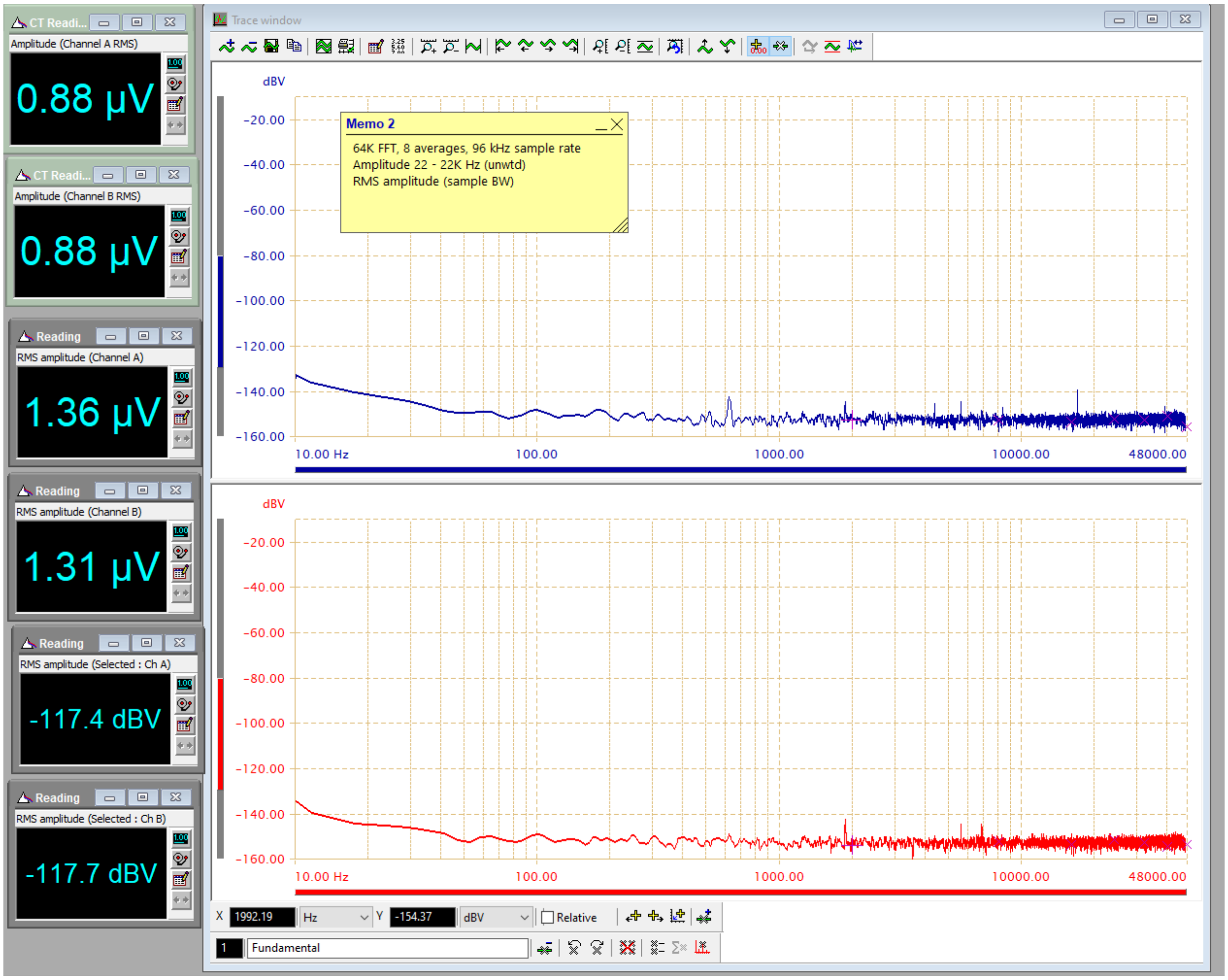

Figure 9 dScope M1 with input shorted, measured 29-Mar-23

The type of variability detailed in the Appendix can be reduced with dedicated test rooms and leaving the equipment permanently installed. Most times this isn't practical and we need to make measurements at an engineering bench.

In the case of testing the AK4619 performance at a 48 kHz sample rate is the focus as that is the typical rate used by A^2B applications. To ensure our measurements extend out to the 24 kHz Nyquist rate the preference is to operate the M1 with an (analog input) sample rate of 96 kHz.

Keep that sample rate difference in mind if you compare unweighted numbers and they were not measured with the same bandwidth, i.e., the ADC at 48 kHz, and the DAC at 96 kHz sample rates.

A-weighted numbers for noise or THD+N would nominally be expected to be the same regardless of the measurement's sample rate as the weighting filter limits the bandwidth considerably.

3 AK4619 PERFORMANCE

The datasheet for the AK4619 specifies typical values (worst case, when specified, ranged from 5 dB to about 10 dB worse than the typical value on the datasheet) is summarized, along with direct (i.e. no A^2B interface) EVM measurements:

- SNR (A wtd, 48 kHz sampling rate, single ended)
  - ADC 104 dB (measured 104 dB)
  - DAC 108 dB (measured 108 dB)
- Dynamic range (A wtd, -60 dBFS)
  - ADC 95 dB (measured 104 dB)
  - DAC 108 dB (measured 108 dB)
- S/(N+D)[7] (48 kHz sampling rate -1 dBFS signal)
  - ADC 95 dB (measured 95 dB in worst channel)
  - DAC 91 dB (measured 92 dB)

In the EVM the DAC clock is coming from a SPDIF receiver, and SPDIF is well known for causing clock jitter unless the SPDIF Rx implements a decent clock/PLL management scheme.

All of the AK4619 performance numbers are > 10 dB worse than the M1's capability so ignoring a setup error the small variations in the M1 instrument limits will not be of concern in measuring the AK4619.

3.1 BASELINE PERFORMANCE OF THE AK4619 EVM MEASURED WITH THE M1

The AK4619 0 dBFS output and input values are 1V RMS make it convenient to display levels in dBV. In the next screen captures there are two levels, unweighted 22 kHz BW (upper two measurements) and unweighted across the full measurement bandwidth (48 kHz).

Other values were obtained by changing the M1 settings to produce results comparable to the datasheet.

[7] We'll treat this the same as THD+N even though there's a very slight difference in the bandwidth as the AKM datasheet would be for 24 kHz and because we're capturing at 96 kHz we use the M1's 22 kHz filter option.

Unless noted otherwise data is shown for channels 1 and 2. No meaningful difference was found between the 4 channels on the AK4619.

3.1.1 DAC MEASUREMENTS

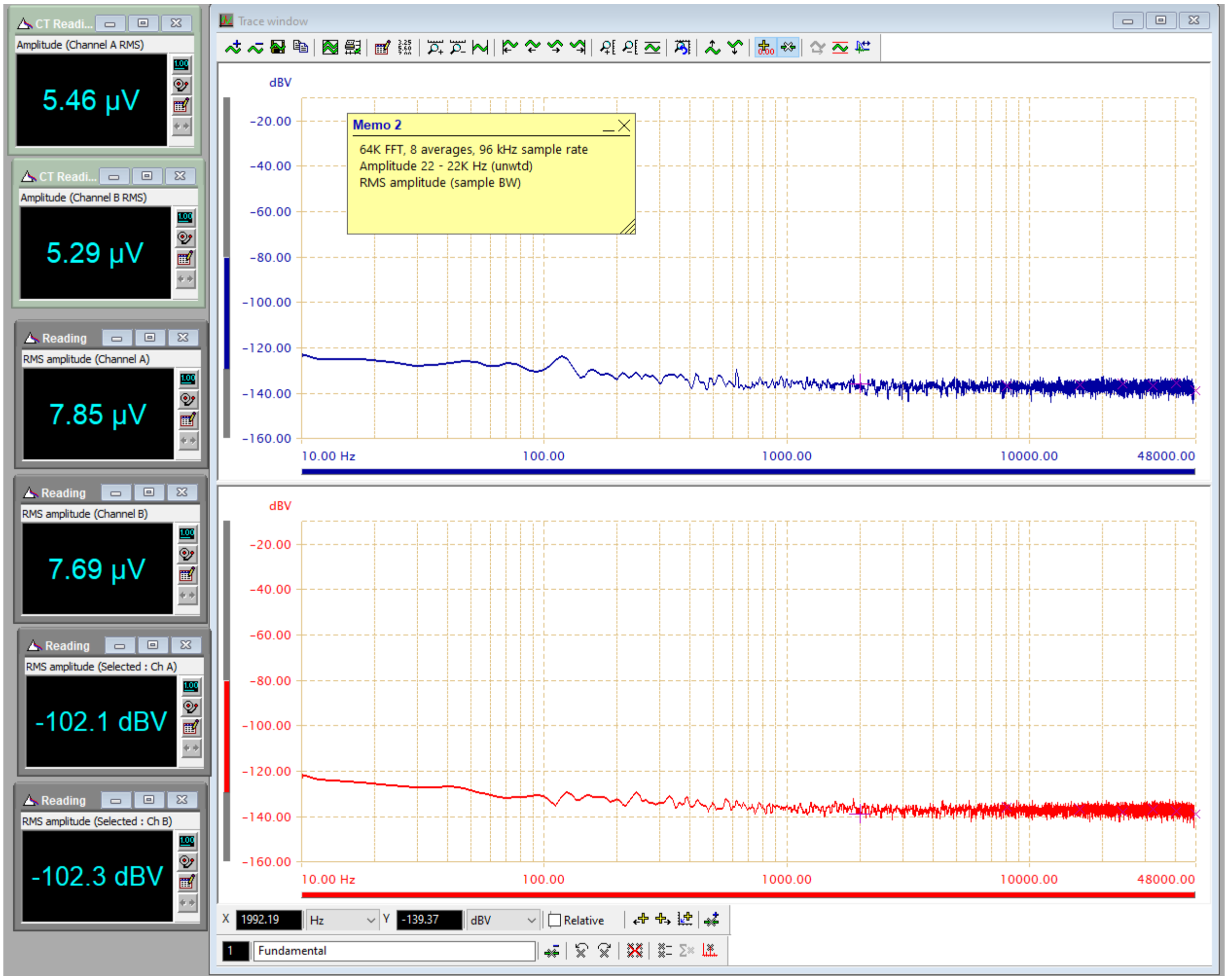

Figure 10 Ch 1 and 2 DAC out with zero input, A-wtd and 96 kHz amplitudes. 22 kHz BW RMS noise is -105 dBV. A wtd (not shown) is 4.0 uV, or -108 dBV.

Distortion with full scale output is dominated by the 3rd harmonic output level of -94.7 dB, Figure 11

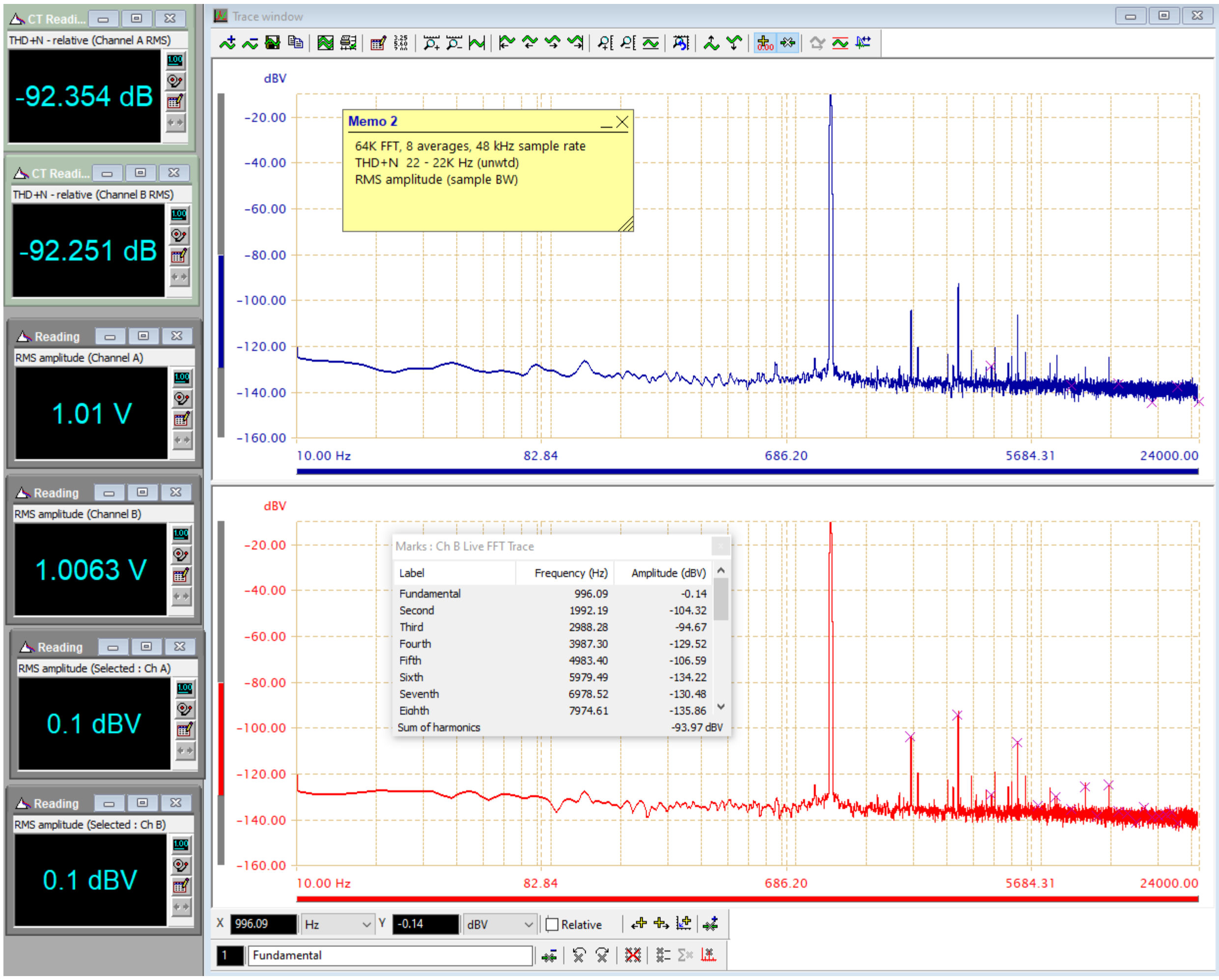

Figure 11 Ch 1 and 2 full scale DAC output

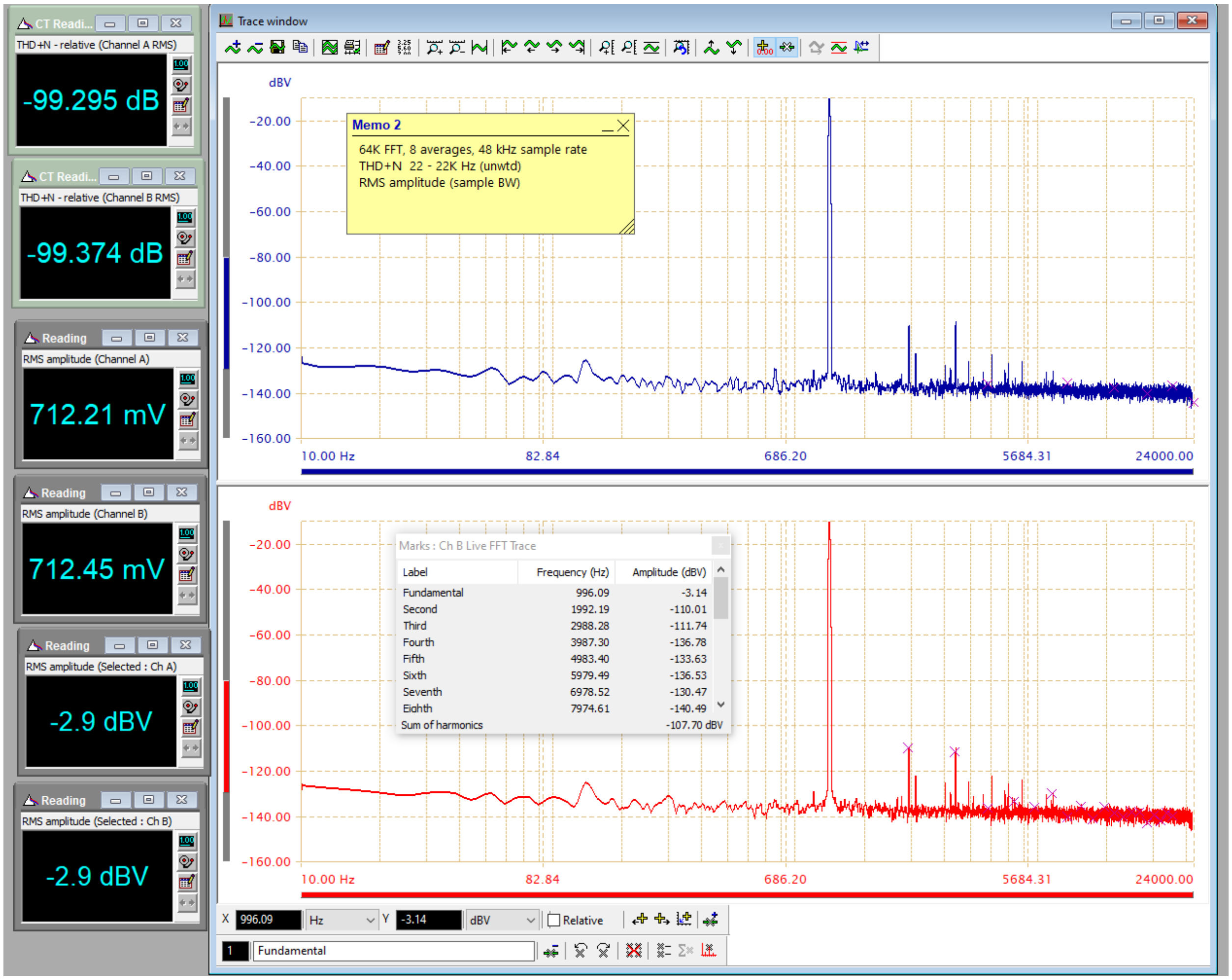

Figure 12 DAC output for -3 dBFS level for chan 1 and 2

For the -60 dBFS test case the DA output shows no harmonic components and the THD+N is dominated by noise. - 48 dB was measured (A wtd), or a DR -108 dB.

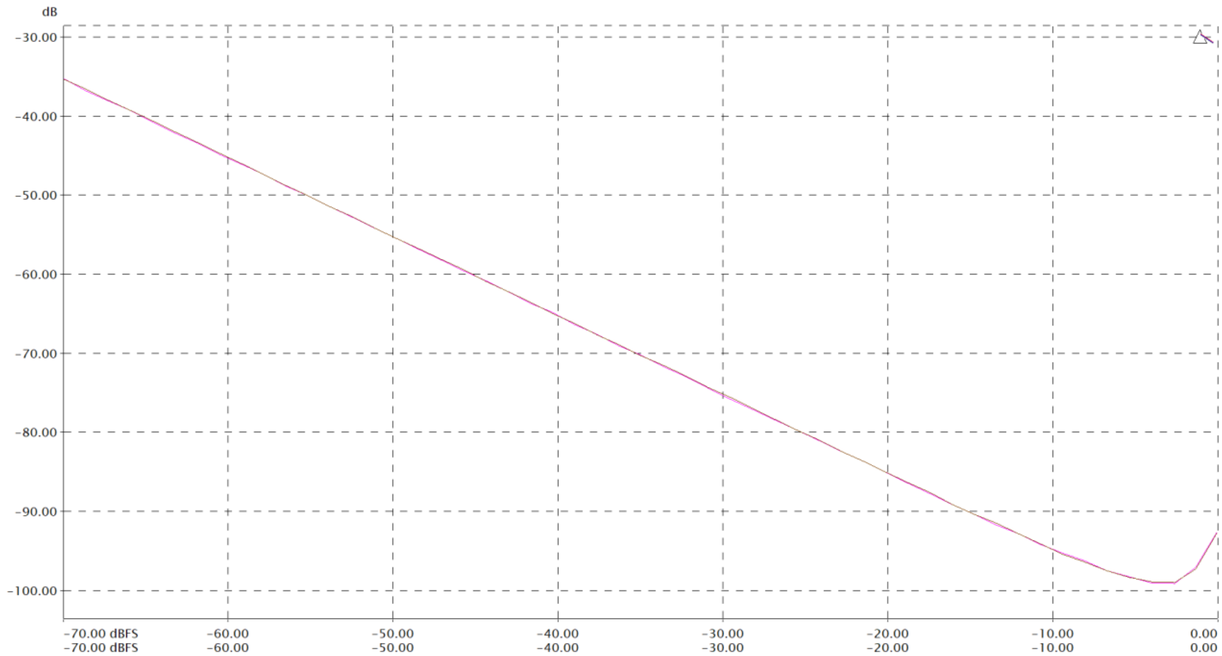

Figure 13 DAC chan 1 and 2 THD+N (unweighted) versus level

3.1.2 ADC MEASUREMENTS

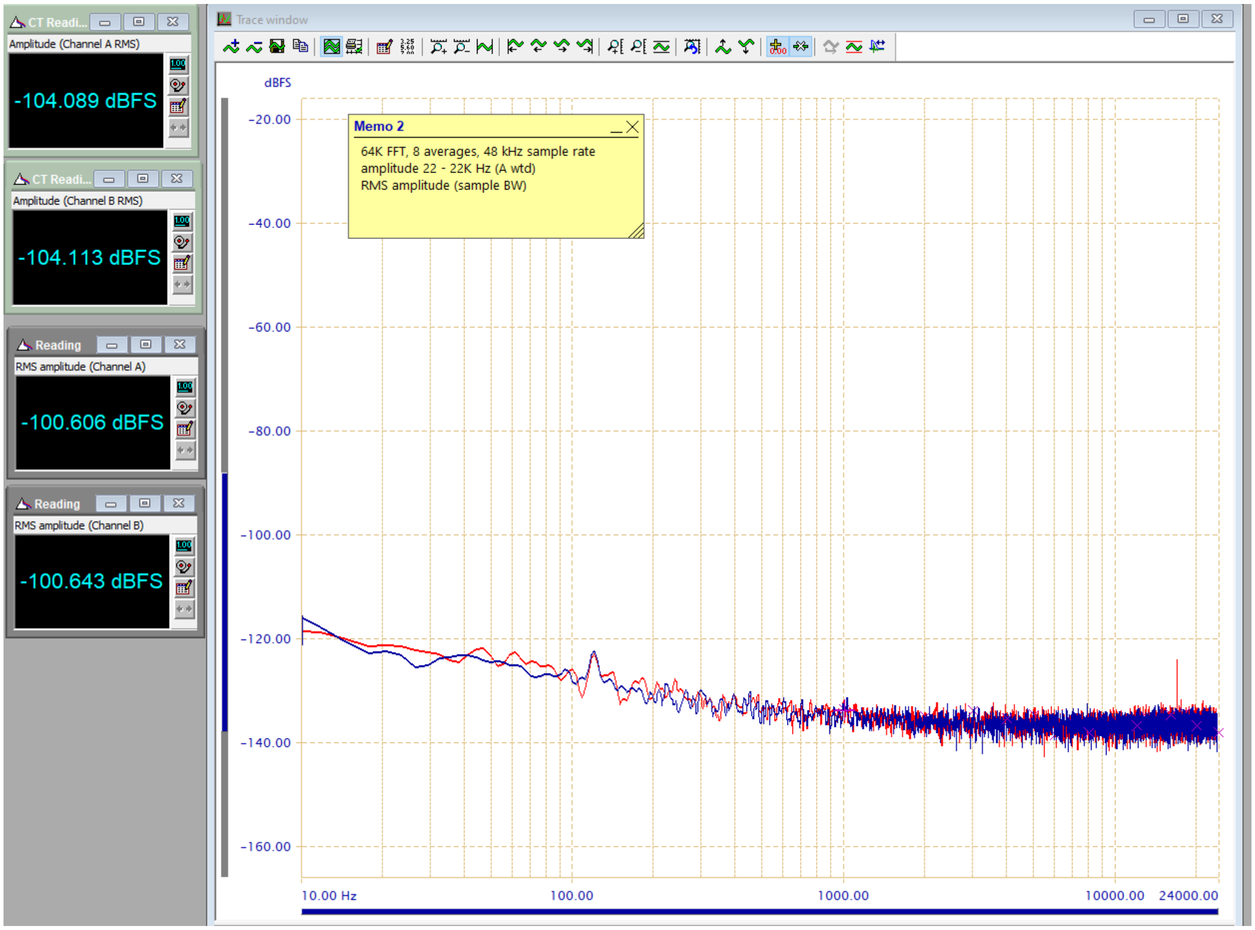

Figure 14 ch1 and 2 ADC output with input shorted.

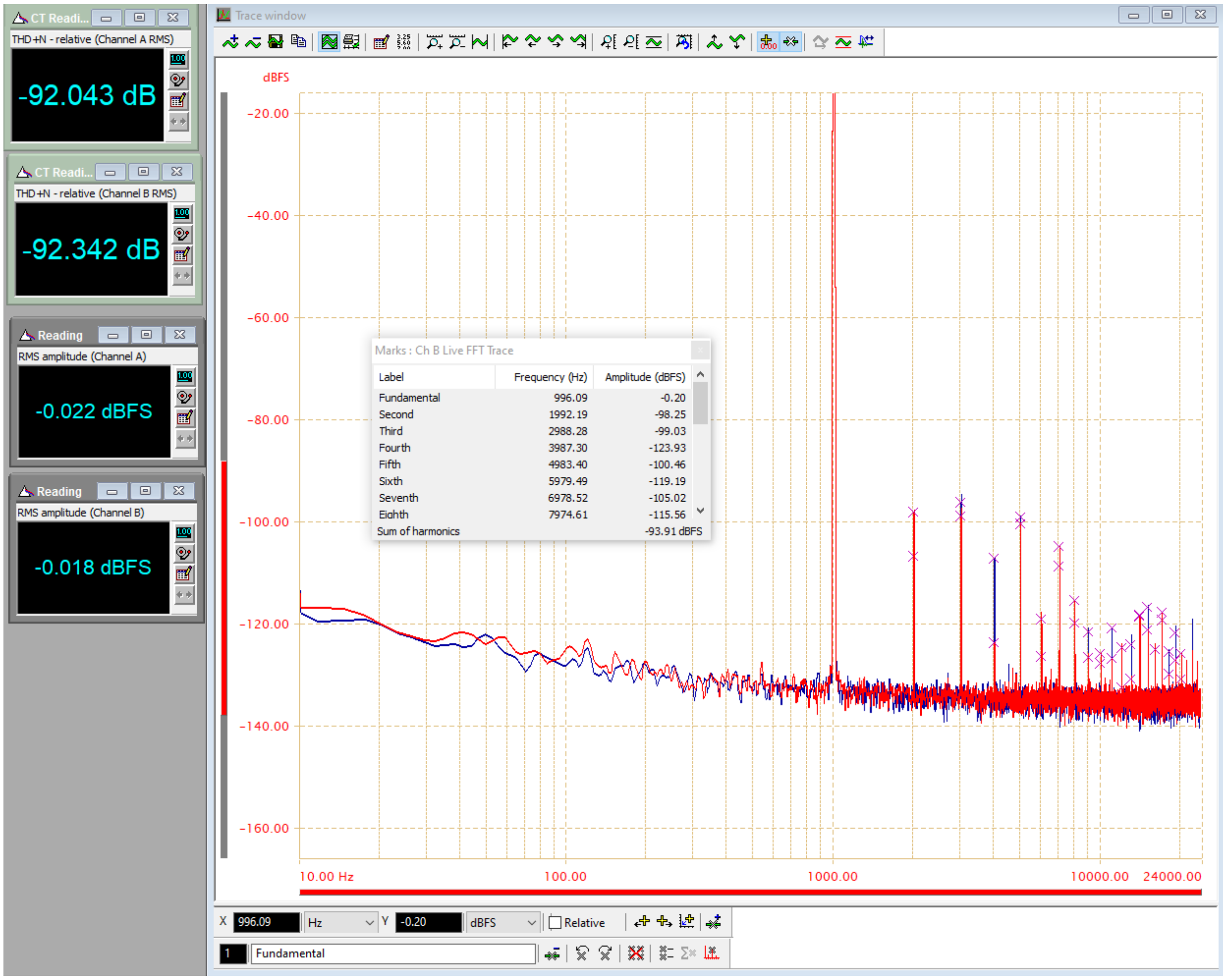

Figure 15 ADC output with full scale input. With −1 dBFS level THD+N drops to −96 dB

The ADC measurement for a −60 dBFS output is −44 dB, which in the context of the way the datasheet describes DR for the ADC, would be 104 dB, which is considerably better than the datasheet number.

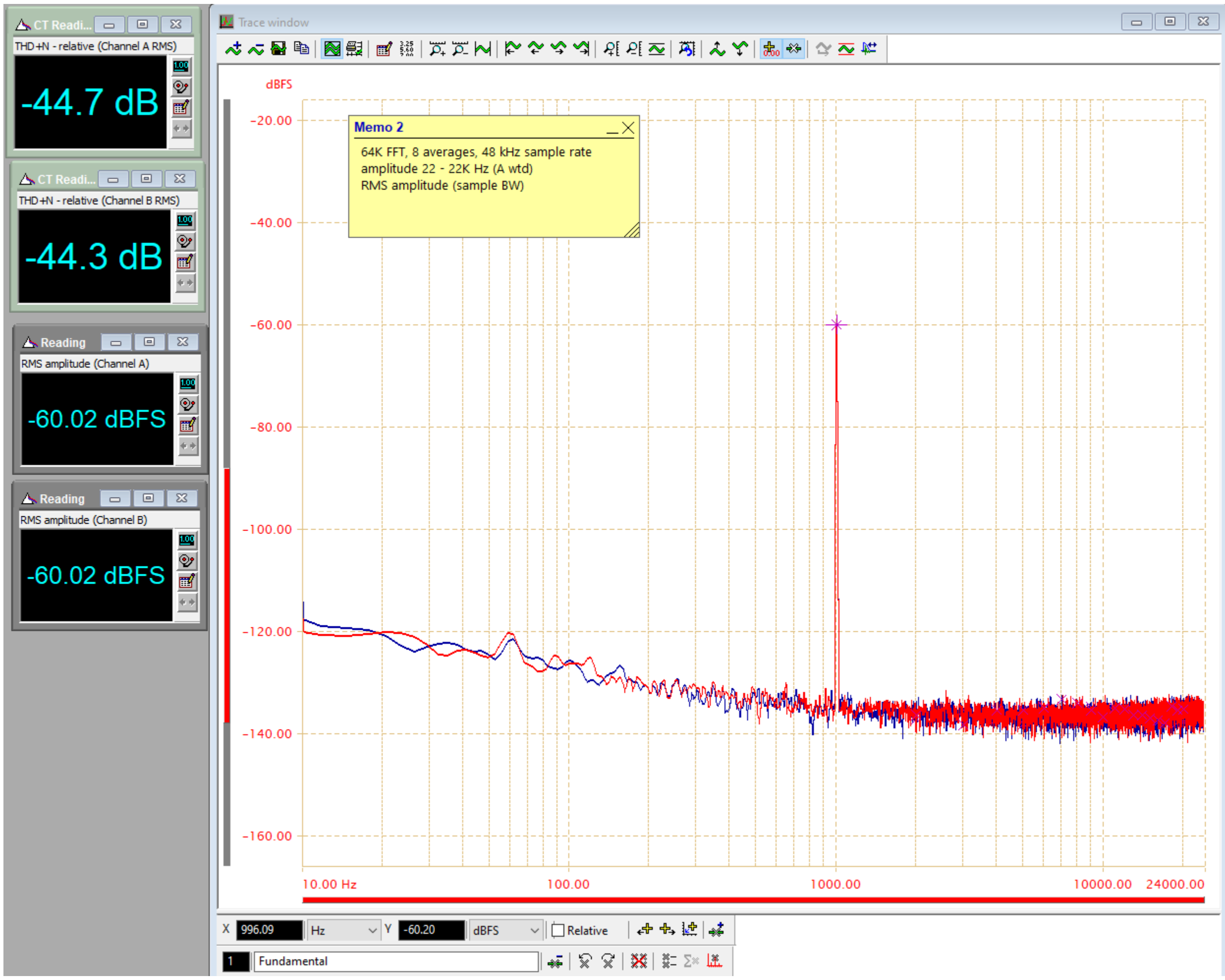

Figure 16 FFT results for input level that produces -60 dBFS output from ADC. No harmonics observed.

Other measurements were made to ensure that anomalies were not overlooked. Since the AK4619 datasheet does not supply graphical data, the measured results are supplied here in the figures below.

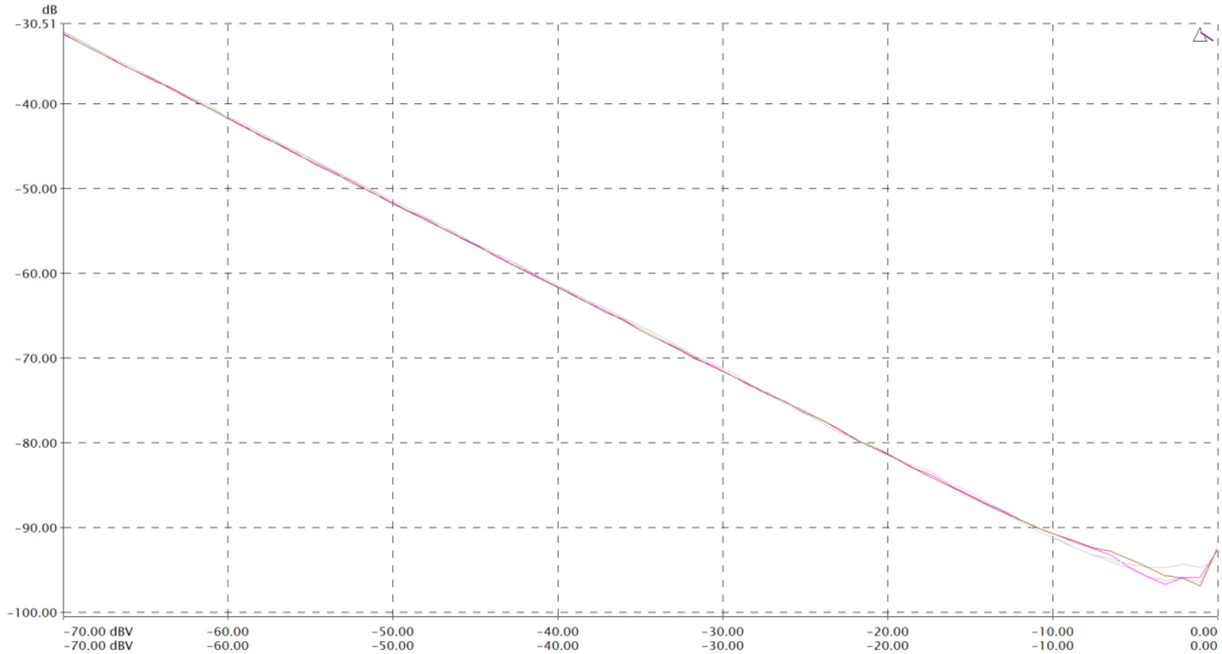

Figure 17 Ch 1-4 ADC THD+N (unweighted) vs. level plot

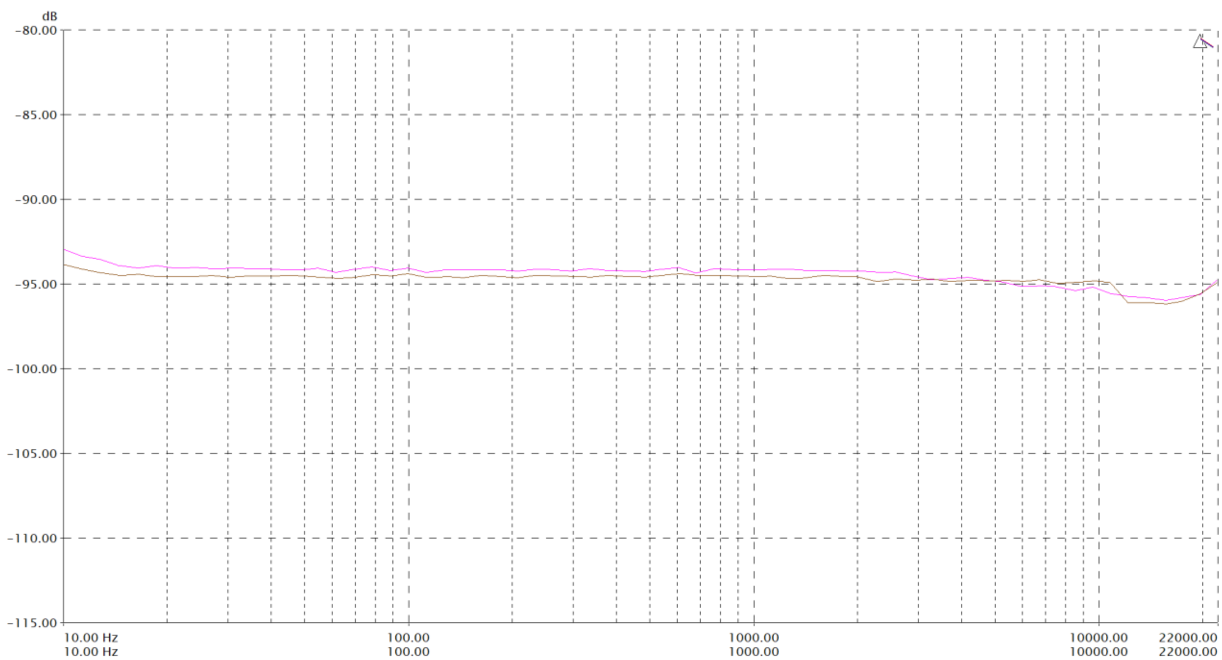

Figure 18 THD+N (unweighted) vs frequency (-6 dBFS signal level)

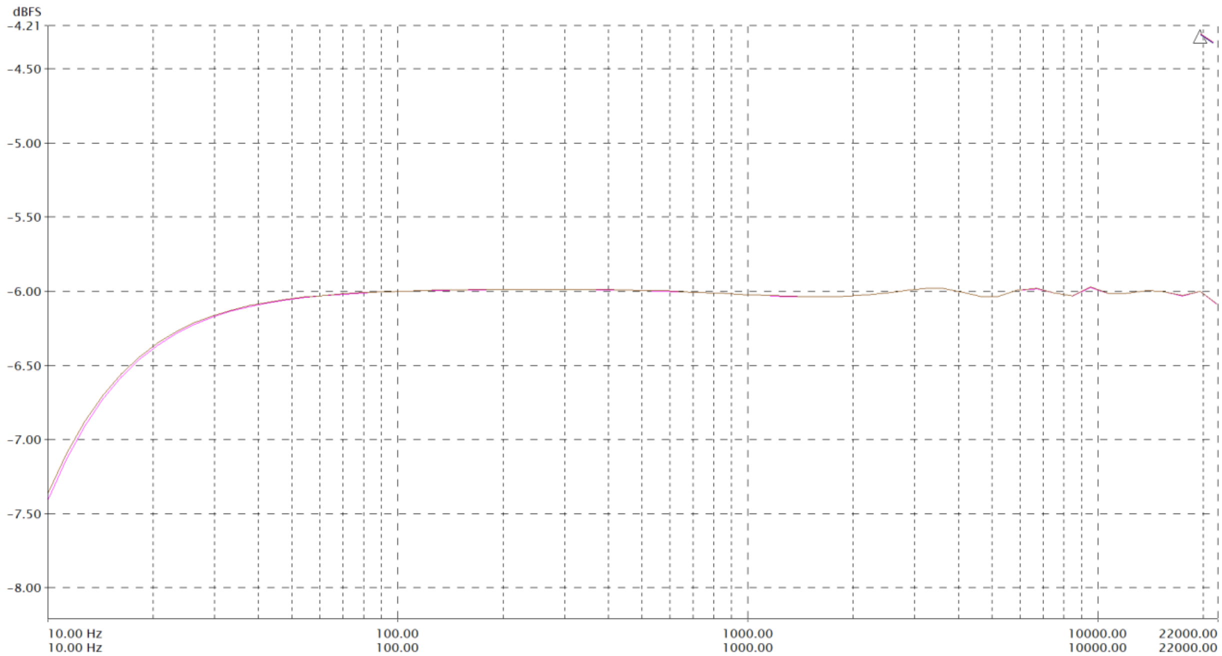

Figure 19 ch 1 and 2, amplitude vs. frequency. The low frequency rolloff is from a combination of the EVM design and the HPF enabled on the M1 to avoid DC offsets from affecting measurements.

3.1.3 WHAT WASN'T MEASURED

There's a lot of other performance measurements that can be made, but for the investigation of any jitter impact the noise floor and harmonic distortion tests are all that are needed. The AK4619 doesn't need input or output buffers – there's not a lot of options to degrade what the chip inherently does.

4 A2B SYSTEM + EVM MEASUREMENTS

The AK4619 EVM (locally clocked) measurements will be used as the baseline to compare the system performance using an A²B interface as the I²S clock source. For testing the DAC the data must be sourced over the A²B bus so that the digital (PCM) data is in the same clock domain as the A²B bus. The WDZ board used as a root node includes an ADAU1452 SigmaDSP and performs the ASRC needed to go from the M1's digital (Toslink) output to the A²B bus's clock domain, see Figure 20. The ADC data is also routed over A²B but is already in the same clock domain so can just be output via Toslink on the WDZ board to the M1 analyzer input.

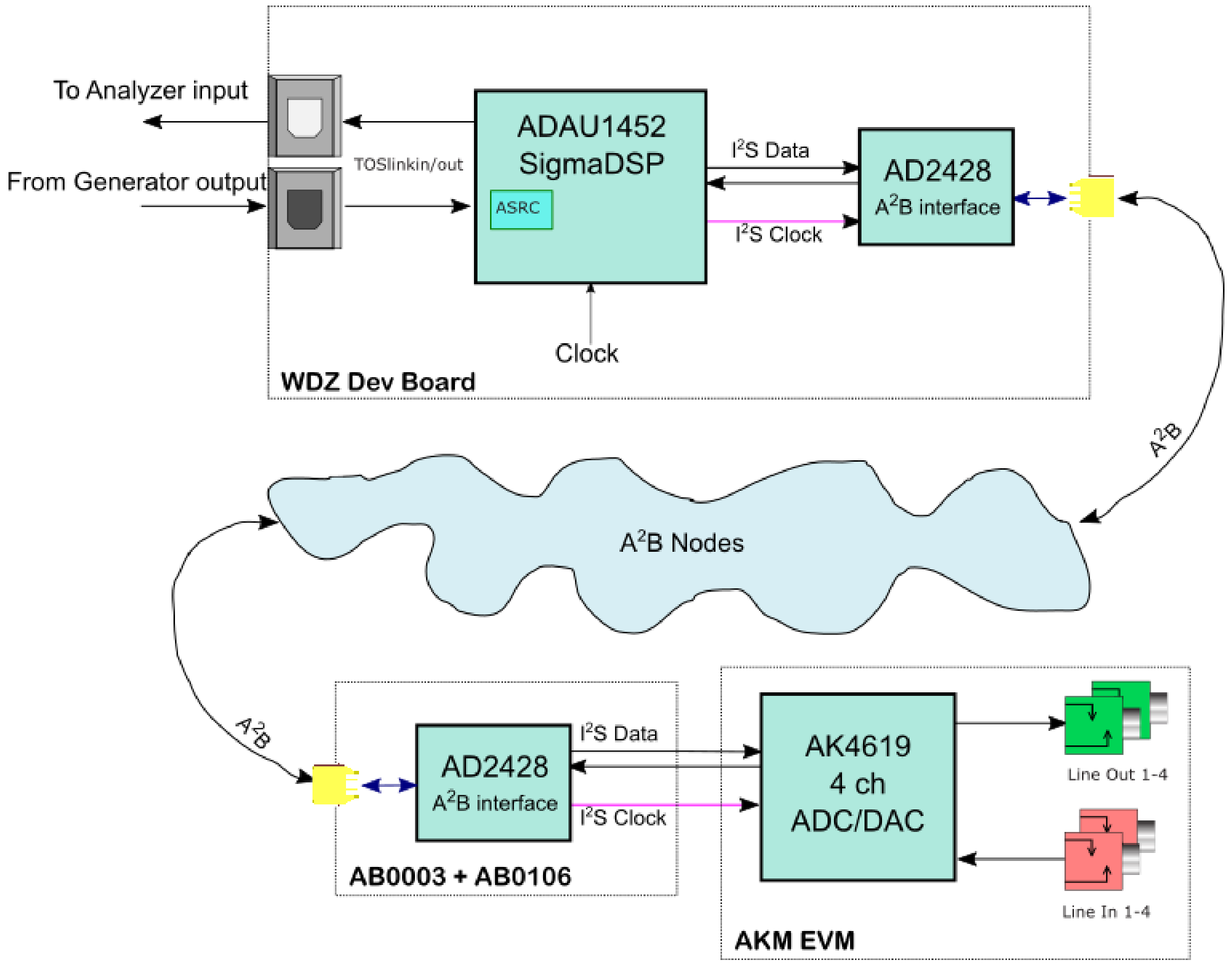

Figure 20 System setup for measuring impact of A²B clock jitter on ADC & DAC performance.

4.1 ADC AND DAC THD+N VERSUS NODE POSITION

The Clockworks measurements of THD and THD+N are presented in the next table, Table 1. Detailed plots are provided after the table. The 4th column is the TIE (Time Interval Error) from the AD2428 datasheet.

The THD value is computed by the dScope M1 summing the level of the first 7 harmonics. THD+N is computed as normal, i.e., removing the fundamental and measuring the resulting signal's amplitude.

Table 1 THD (2nd – 8th) and A weighted THD+N measurement summary. Channel 1, 997Hz, -3 dBFS level

Node	ADC THD measurement	ADC THD+N measurement	ADI AD2428 TIE RMS	DAC THD measurement	DAC THD+N measurement
Direct (no A2B)	-104.5 dB	-96.3 dB	-	-107.7 dB	-99.3 dB
1st	-104.8 dB	-96.3 dB	1.57 nsec	-109.2 dB	-98.6 dB
5th			2.15	-109.1 dB	-96.6 dB
9th	-104.5 dB	-94.4 dB	2.58	-109.3 dB	-90.4 dB

From the table we see that the AK4619 is susceptible to the large jitter amounts that can be present in an A^2B system. Given the native/ideal performance of the AK4619 this relative loss in performance is relatively minor for the ADC portion and probably is not a concern for the DAC either, despite the 9 dB increase in THD+N.

The noise floor was not impacted by the jitter. For the ADC the A weighted noise was measured as -104 dBFS and that same value is observed with the test node at location 1 or 9.

4.2 SPECTRUM

The THD and THD+N numbers provide a general sense of where issues may be occurring. As a single number they don't provide insight into the details. For a better understanding the spectrum of the AK4619 EVM output was captured.

4.2.1 ADC OUTPUT AT A^2B NODE 1

Looking at Figure 21 notice the increase in the noise floor around the fundamental compared to just the EVM as shown in Figure 15. The increase is not large enough to change the THD+N value in Table 1.

It's also noted that the harmonics for channel 2 have a different pattern than channel 1, which is somewhat unexpected. Regardless, the noise floor is dominating the THD+N value.

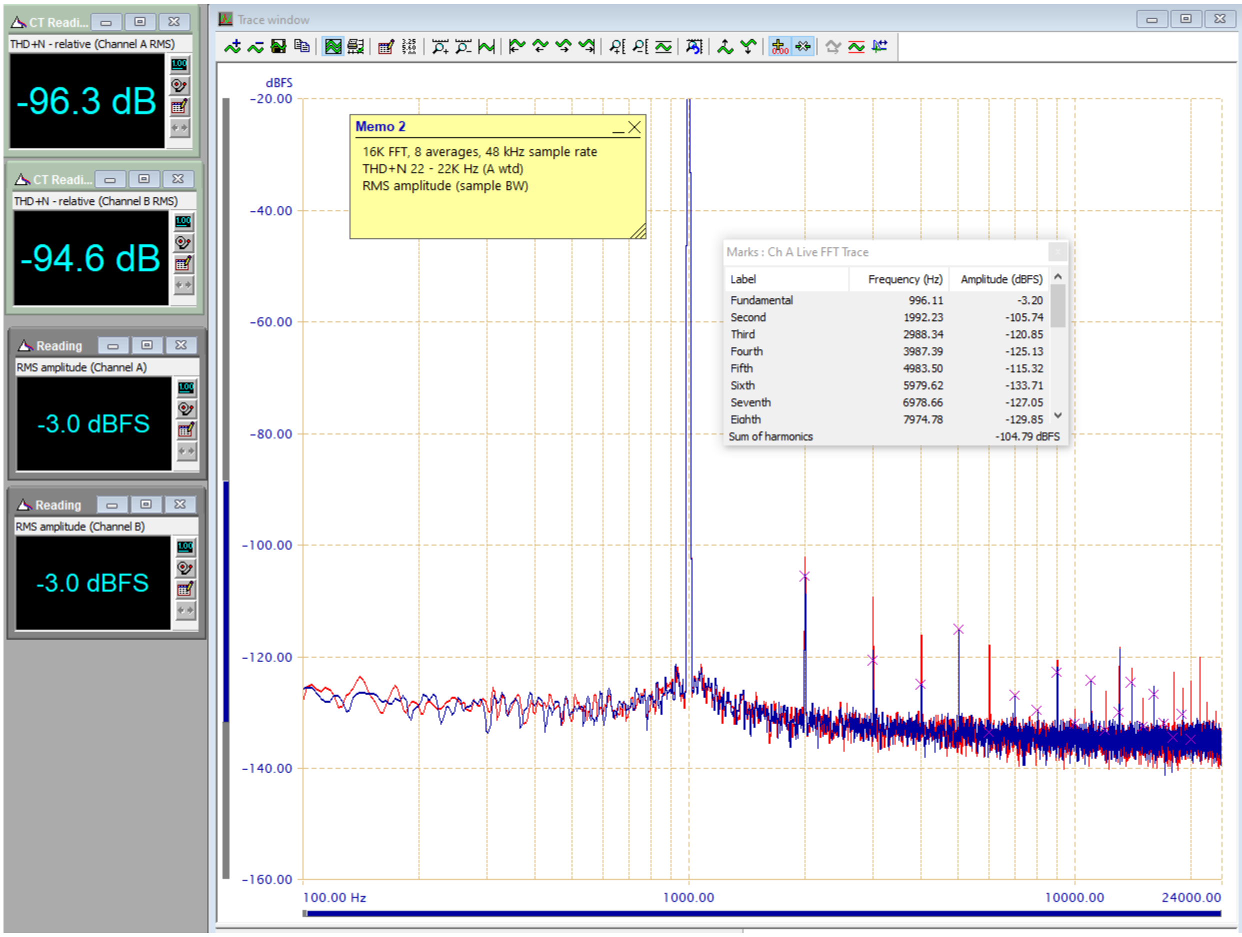

Figure 21 ADC output with 0.5V RMS input corresponding to -3 dBFS for EVM operated as the first downstream node.

4.2.2 ADC OUTPUT AT A²B NODE 9

The increase in the noise floor and subsequent degradation of THD+N is more obvious when we use the 9th node. Looking at Figure 22 and comparing the noise floor to Figure 21 it's obvious there's been an increase. Figure 23 shows a higher resolution view around the fundamental to show the increased noise floor.

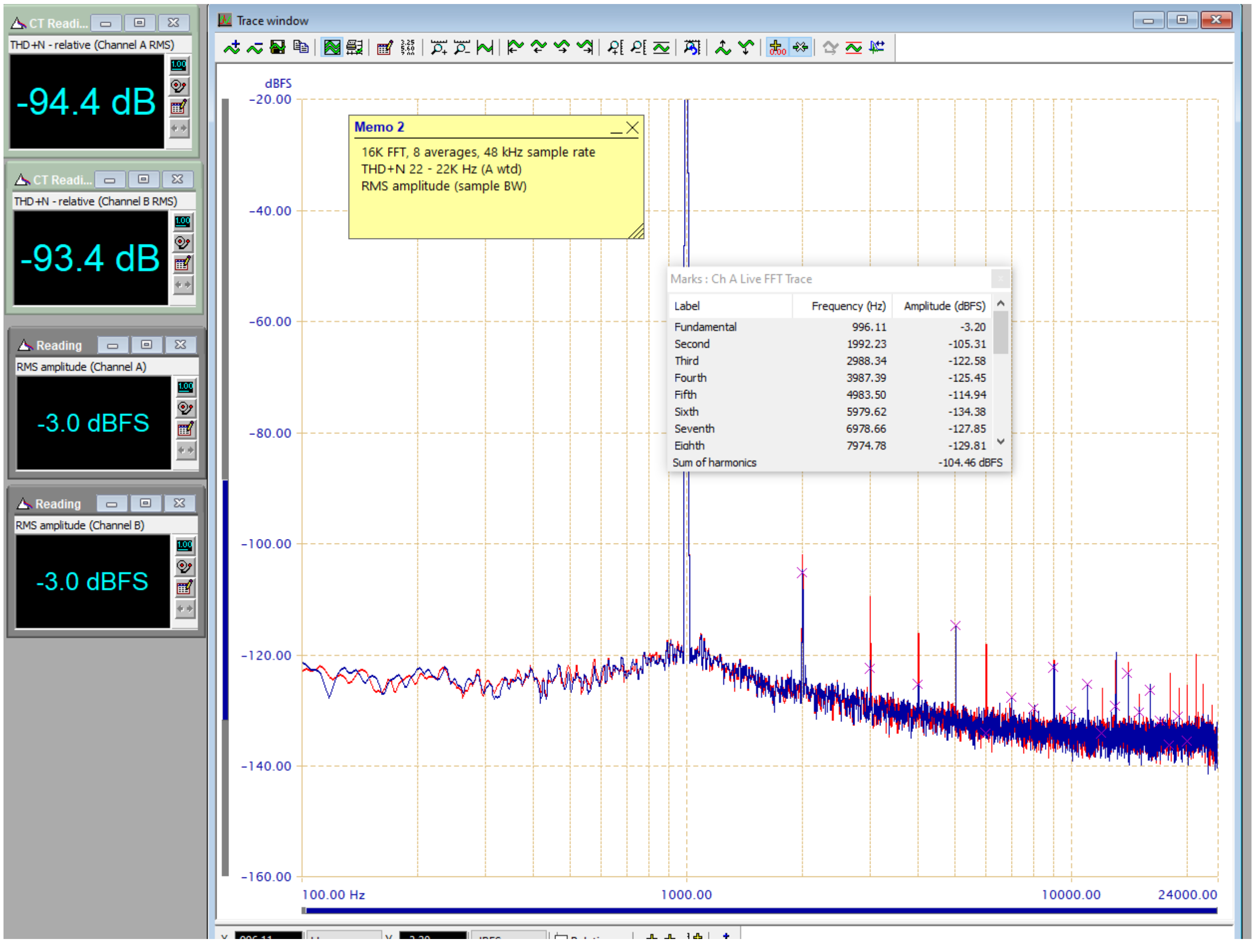

Figure 22 ADC output with 0.5V RMS input corresponding to -3 dBFS for EVM operated as the 9th downstream node.

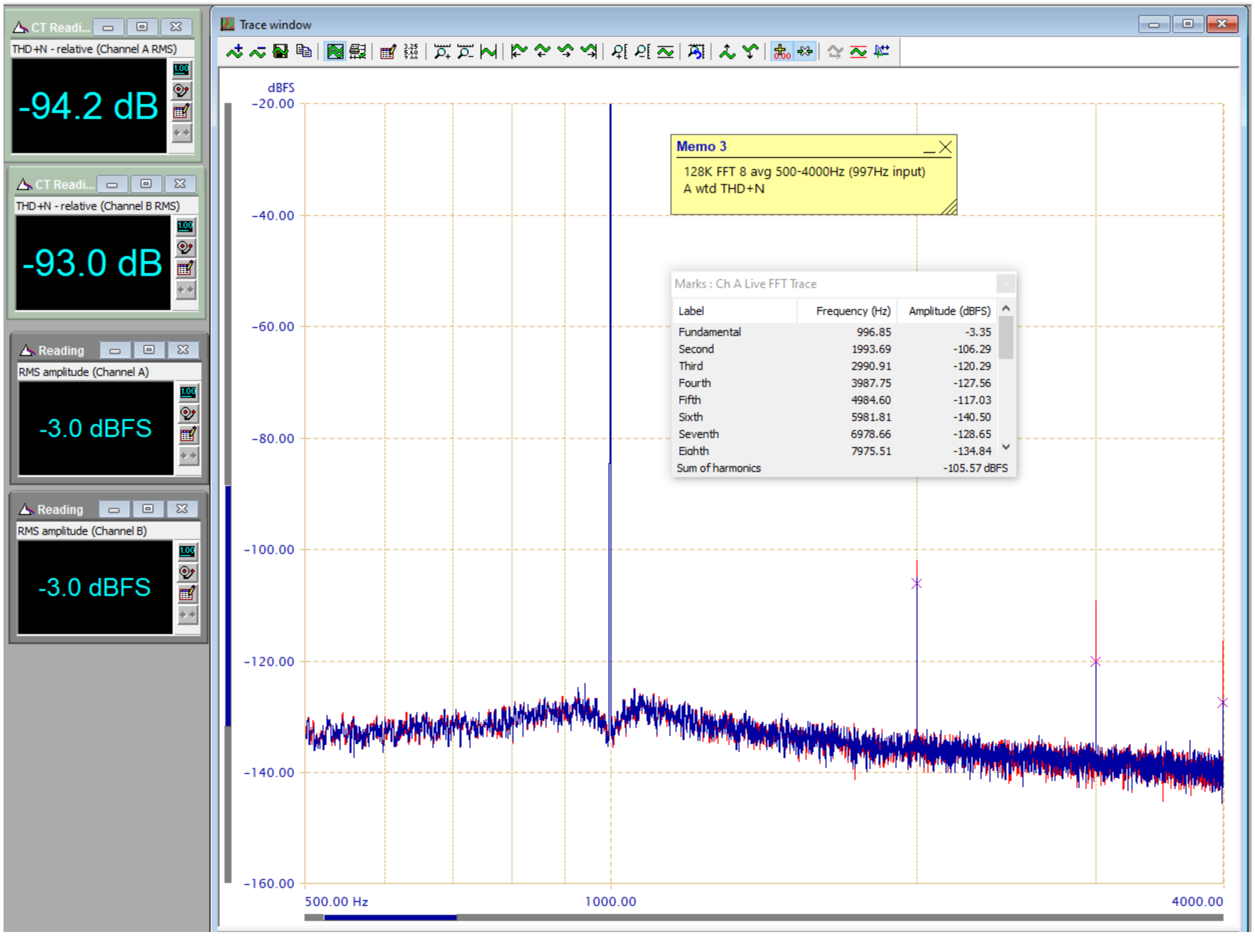

Figure 23 High resolution FFT of ADC output with -3 dBFS input and operated as the 9th downstream node.

Figure 24 summarizes the THD+N performance between the lower jitter node 1 location and the node 9 location. In the harmonic distortion dominated part of the curve, -10 dBV to 0 dBV in, the higher jitter node 9 degrades the ADC THD+N performance by a few dB.

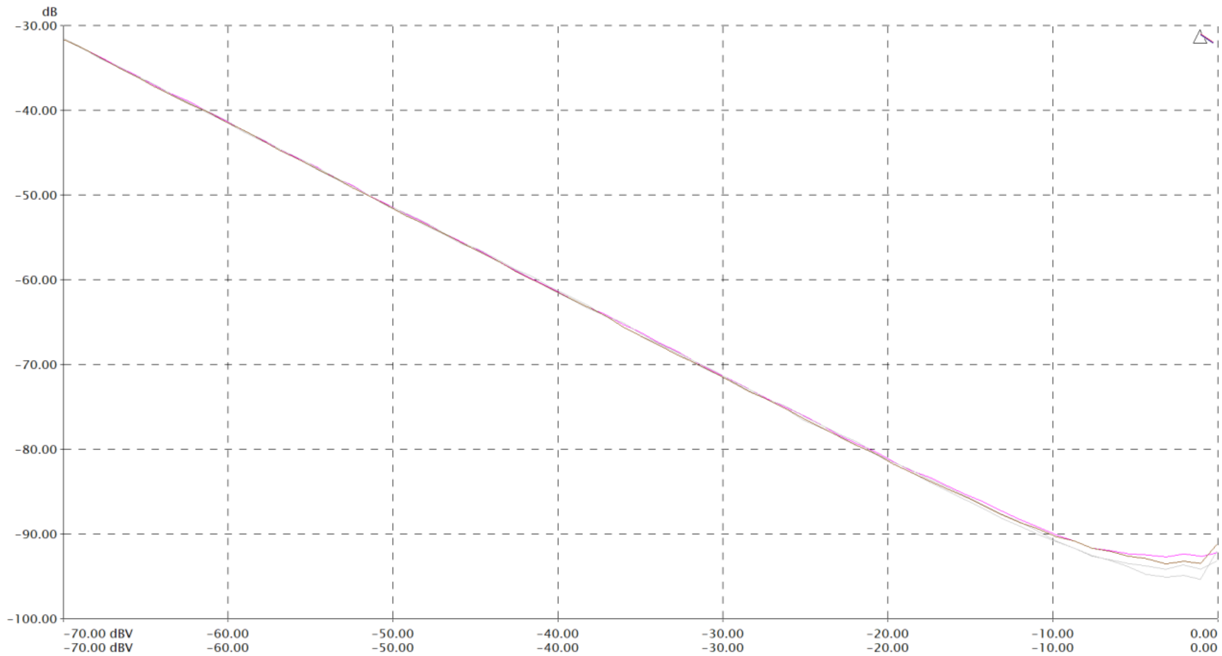

Figure 24 Comparison of ADC THD+N vs. Level for node 1 (bottom two traces) vs node 9 (upper two traces)

4.2.3 DAC OUTPUT AT NODE 1

The lowest output distortion occurs around the – 3dBFS level. Figure 11 Figure 12 is the DAC output for the plain EVM; compare that to Figure 25 where the EVM is operating from the first downstream A^2B node location. The THD+N has degraded by about 0.7 dB, though we see the THD is actually a little better. With the low harmonic levels there is some uncertainty in the exact levels so the unexpected change has other possible explanations.

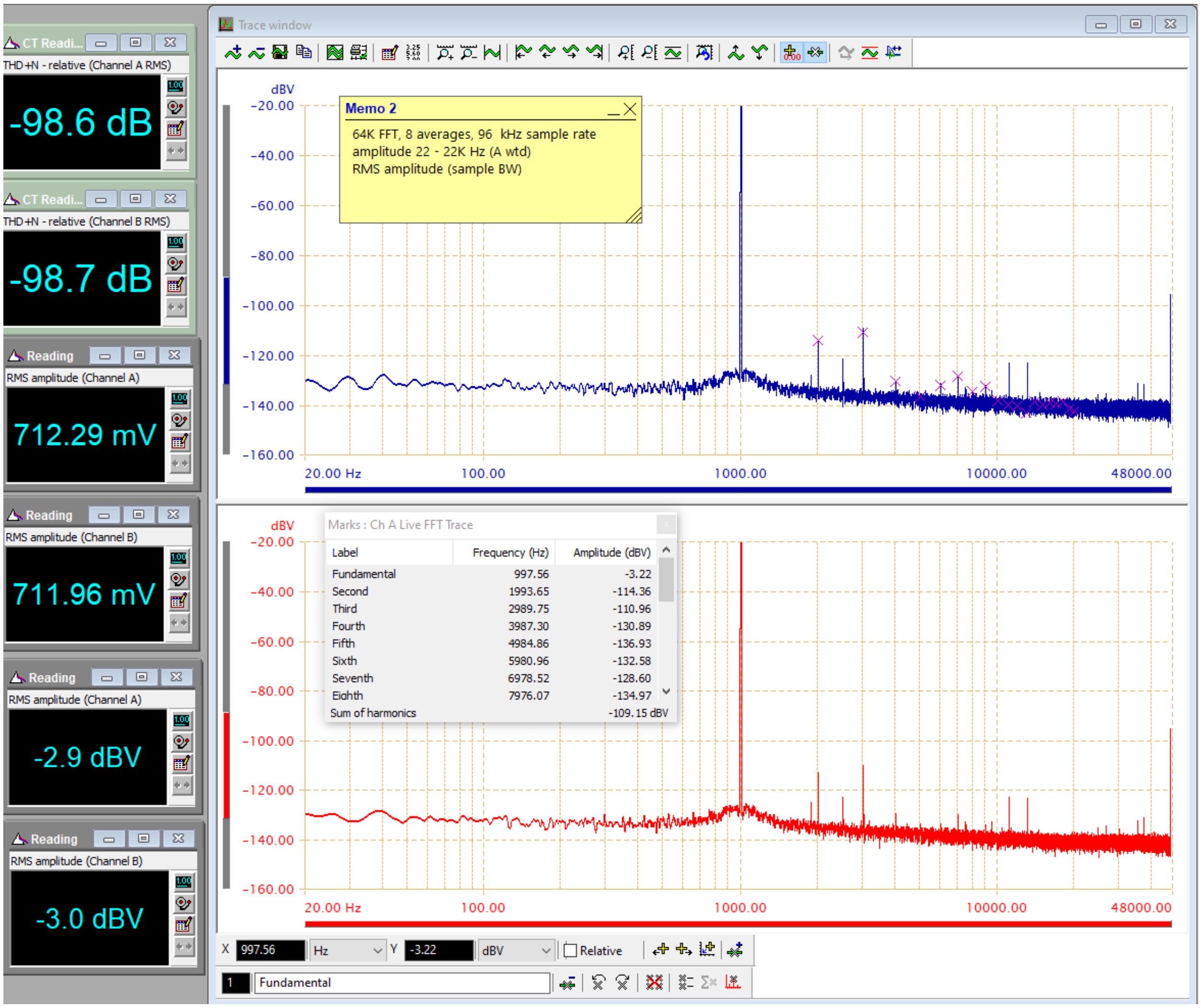

Figure 25 chan 1 and 2 DAC output for node 1 location

4.2.4 DAC OUTPUT AT NODE 9

As shown in Figure 26, at node 9 the effect of the increased jitter has manifested itself more clearly. There’s a definite skirt around the fundamental and THD+N has dropped 3.5 dB while the THD level has remained about the same as seen in the node 1 case.

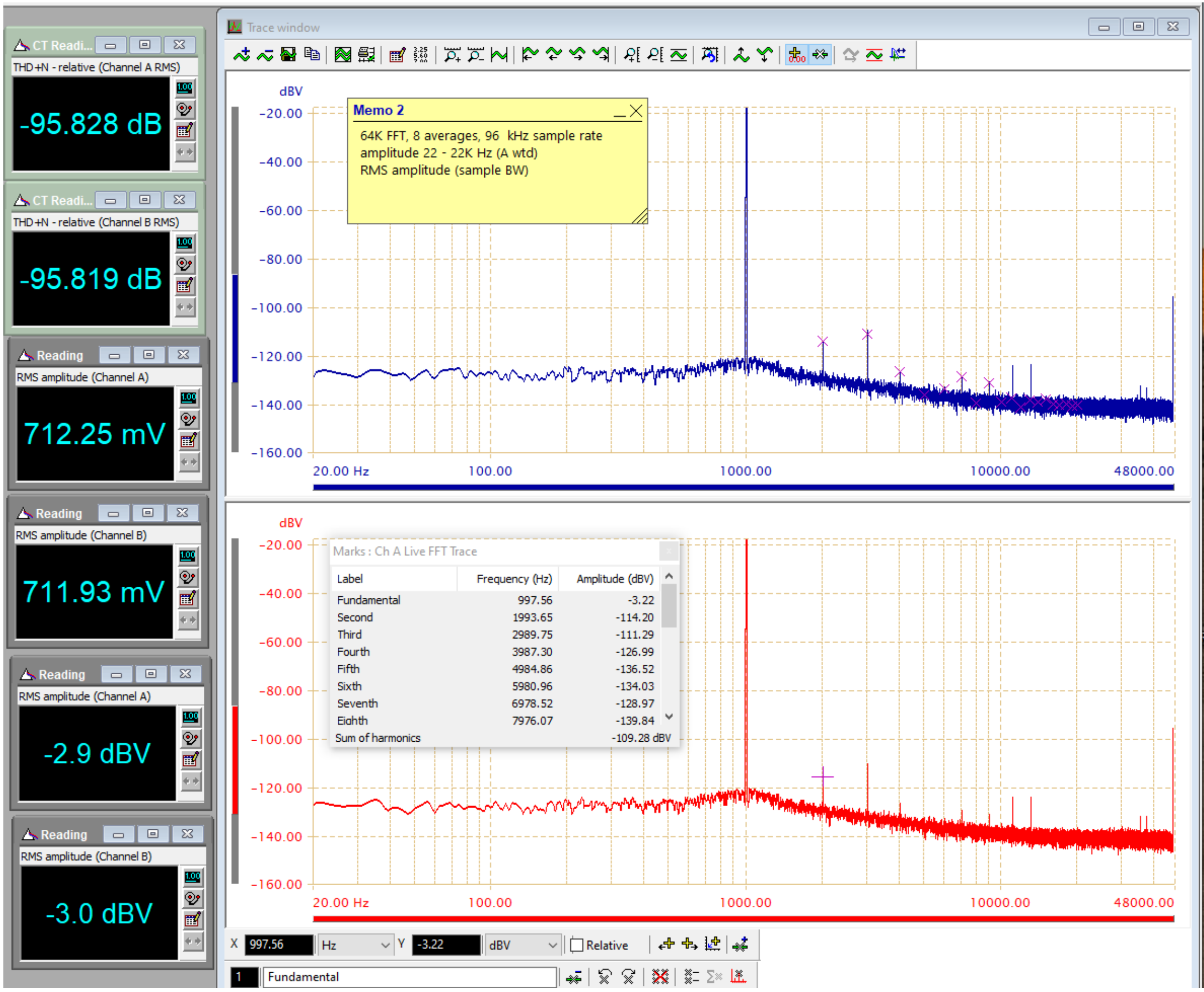

Figure 26 Chan 1 and 2 DAC output for node 9 location

Comparing the THD+N versus level (unweighted) plots of just the EVM (Figure 13) with the same test on node 9 (Figure 27) we see that the THD+N curve for the EVM case remains flat until about −6 dBFS. By comparison the THD+N curve for the DAC operating at node 9 starts breaking around −12 dBFS. With the THD levels remaining the same percentage of the fundamental then the increase must be from the noise (N).

Broadband signal level dependent noise is a typical side effect of high clock jitter.

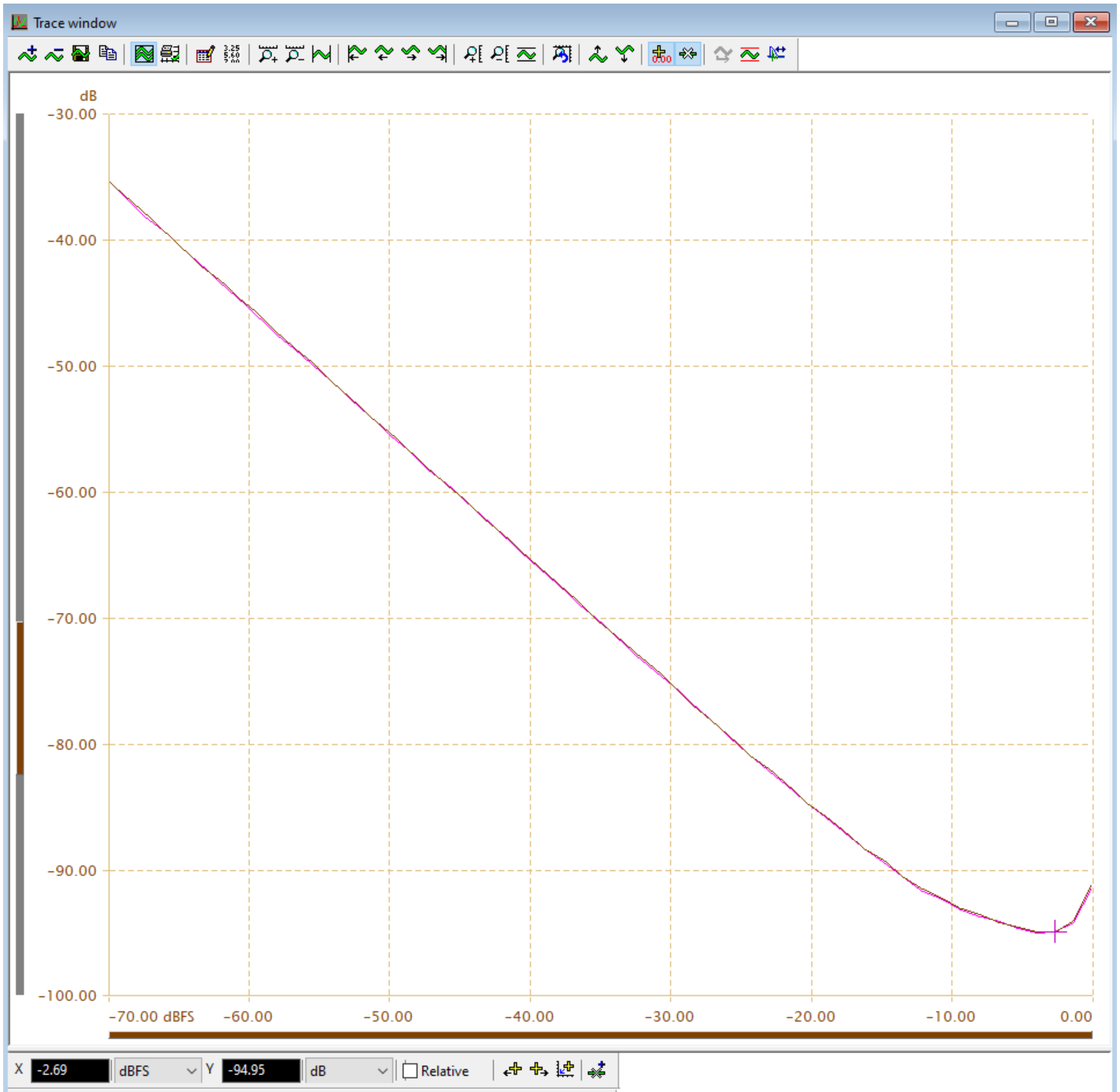

Figure 27 DAC chan 1 and 2 THD+N (unweighted) vs. level plot (997 Hz). Compare with Figure 13.

5 WHAT TO DO?

There are three choices in A^2B system design if the desired DAC or ADC performance degradation from the AD2428's I^2S jitter is problematic:

- Add a clock cleanup chip.
- Add an ASRC
- Pick a different DAC or ADC

There are pros and cons with all three of these possible solutions but are outside the scope of this TechNote. The ASRC is easiest in that there's no need to do further analysis. The first option may need to include some buffering in the hardware design for the I^2S data in case the clock's total error becomes a significant portion of the bit clock period.

Choosing a different ADC or DAC requires making measurements like those described in this document that the part actually does handle jittery clocks; there's no real standard for specifying the details of how a part reacts to all of the possible ways the clock could be jittered.

6 SUMMARY

The I^2S outputs of the AD2428 were previously shown to have high jitter relative to typical audio applications. The testing here demonstrates an AKD4619 EVM measured DAC THD+N of -99 dB can be reduced to -90 dB or less when used in an A^2B network application. It appears as an increased noise level in the presence of large signals versus tones or other narrowband effects that might have increased audibility vs. a broadband noise increase.

The A^2B jitter does not degrade SNR and DR of the AK4619 which from a practical aspect may be more important than THD+N.

If the application is inside of a product that won't have THD+N directly measured (for example a powered speaker) then the THD+N degradation would be inconsequential. In a piece of audio electronics that may be reviewed with high performance test equipment (e.g., same or better than used here) then the THD+N and FFT spectral analysis degradation may be of concern.

APPENDIX A THINGS THAT WENT BUMP IN THE NIGHT

A.1 GREMLINS

As mentioned elsewhere the bench location is sometimes bothered by 60 Hz fields and a nearby radio broadcast tower. Most of the time these can be ignored as the levels involved (< 0.1 uV) do not affect the measurements. When looking at the spectrum the bumps from interference sources may be disconcerting as you can't be sure if the noise might be a problem from the equipment instead of interference. Generally moving things around or even just coming back to a particular test later will resolve the problem.

Sometimes though the problem is more stubborn. Compare this next set of figures (Figure 28 – Figure 30) of the dScope M1 input shorted measurement taken at different times with Figure 9. We can see quite a few more noise spikes. The only other difference was the M1 was moved to a different location and then put back in what should be the same exact spot. It's observed that the A weighted noise floor dropped slightly, and the unweighted floor rose slightly in one channel.

Table 2 summarizes the differences over time between the noise floor measurements, both weighted and unweighted. Channel A shows about a 2.5 dB variation across measurements, and Channel B almost a 5 dB measurement variation. It's only by observing the spectrums do we start to understand the nature of the noise and these variations.

In almost all cases the spikes in the noise spectrum are < 100 nV and do not contribute much energy to the RMS measurements.

Figure 28 dScope M1 with input shorted, measured 7-Apr-23

The author is not aware of any changes in the 6 hours between the data captured in Figure 28 and Figure 29. The noise floor is 2 dB better and the tones in the floor have shifted around.

If we were measuring devices with a SNR > 115 dB we might need to create a much more sophisticated test environment to eliminate environmental related interference. For the AK4619 we can reasonably ignore these issues, other than being sure to measure a baseline periodically.

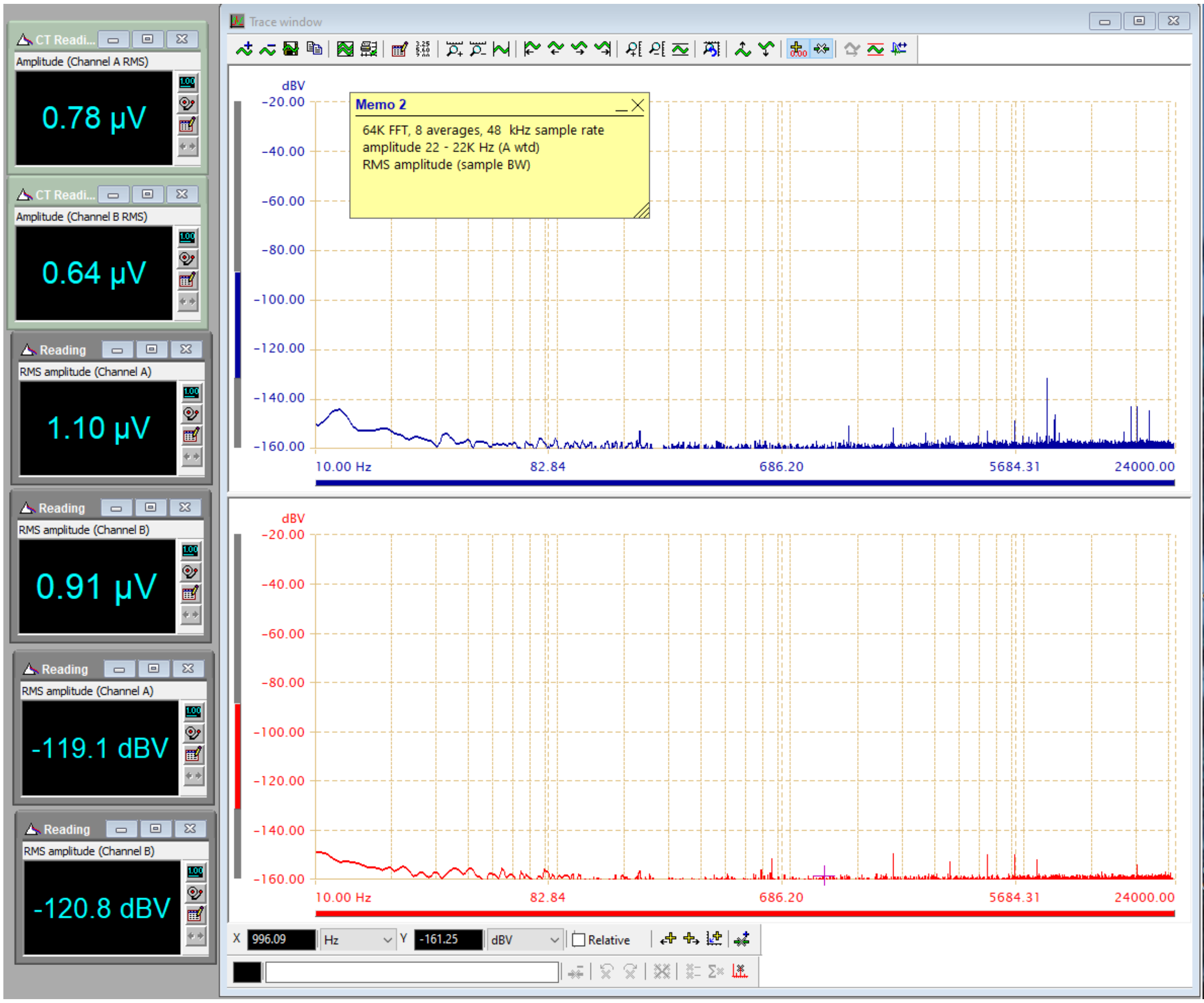

Figure 29 dScope M1 with input shorted, measured 7-Apr-23, about 6 hours after Figure 28 was made

When the noise check was repeated for another data capture a few days later the noise spectrum had changed significantly, as shown in Figure 30.

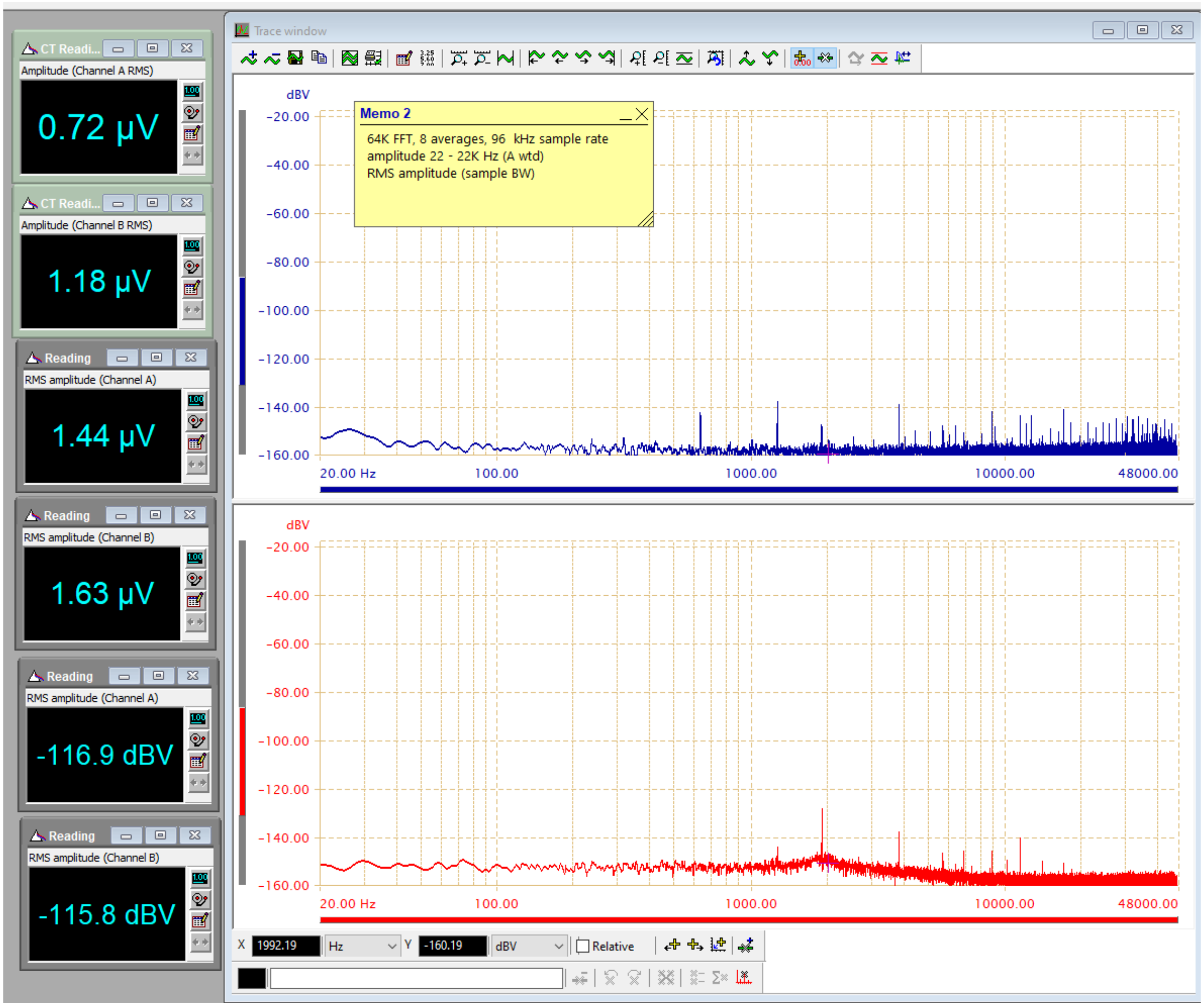

Figure 30 M1 with both inputs shorted, April 11th.

Table 2 Comparison of M1 instrument noise floor on different dates (Left = chan A, right = chan B). The last 2 columns provide the value of the prior 2 columns in dBV.

Capture	A Wtd L	A Wtd R	48K BW L	48K BW R	48K BW L	48K BW R
Figure 9	0.88 uV	0.88 uV	1.36uV	1.31uV	-117.4 dBV	-117.7 dBV
Figure 28	0.68 uV	0.65 uV	1.44uV	1.30uV	-116.9 dBV	-117.7 dBV
Figure 29	0.78 uV	0.64 uV	1.14uV	0.91uV	-119.1 dBV	-120.8 dBV
Figure 30	0.72 uV	1.18 uV	1.44uV	1.63uV	-116.9 dBV	-115.8 dBV

A.2 WATCH OUT FOR FFT PROCESSING GAIN

When performing the measurements sometimes a 16 K point FFT was used to speed up the update rate of the FFT compared to the 64 K point transform size that was generally preferred for capturing data to display in this document.

The smaller bin size (0.73 Hz) of the 64 K point transform better shows pickup of 60 Hz (AC mains in US) noise sources.

Unfortunately, the M1 software does not automatically annotate the captured data (plot+ readings)[8] with the test details, so a note was added by hand. Sometimes it was observed that the FFT size was not set as expected; and while an effort was made to correct the plots a few incorrectly labeled ones may have slipped in to this document.

A.2.1 EFFECT OF FFT SIZE

Since we were plotting amplitude on the Y-axis to determine the harmonic distortion component's values, changing the FFT size will make the noise floor appear lower in the plot as the FFT size in increased. This apparent gain scales as the square root of the FFT size. For our example here the gain = 10 log (4096/2) / log (65536/2) or about 7.3 dB. This is not quite exact because the window function has an effect as well.

[8] This is included when exporting a pdf or other formats for printing, but for this TechNote it was easier to get the desired plot and measurement information using the Windows snipping tool.

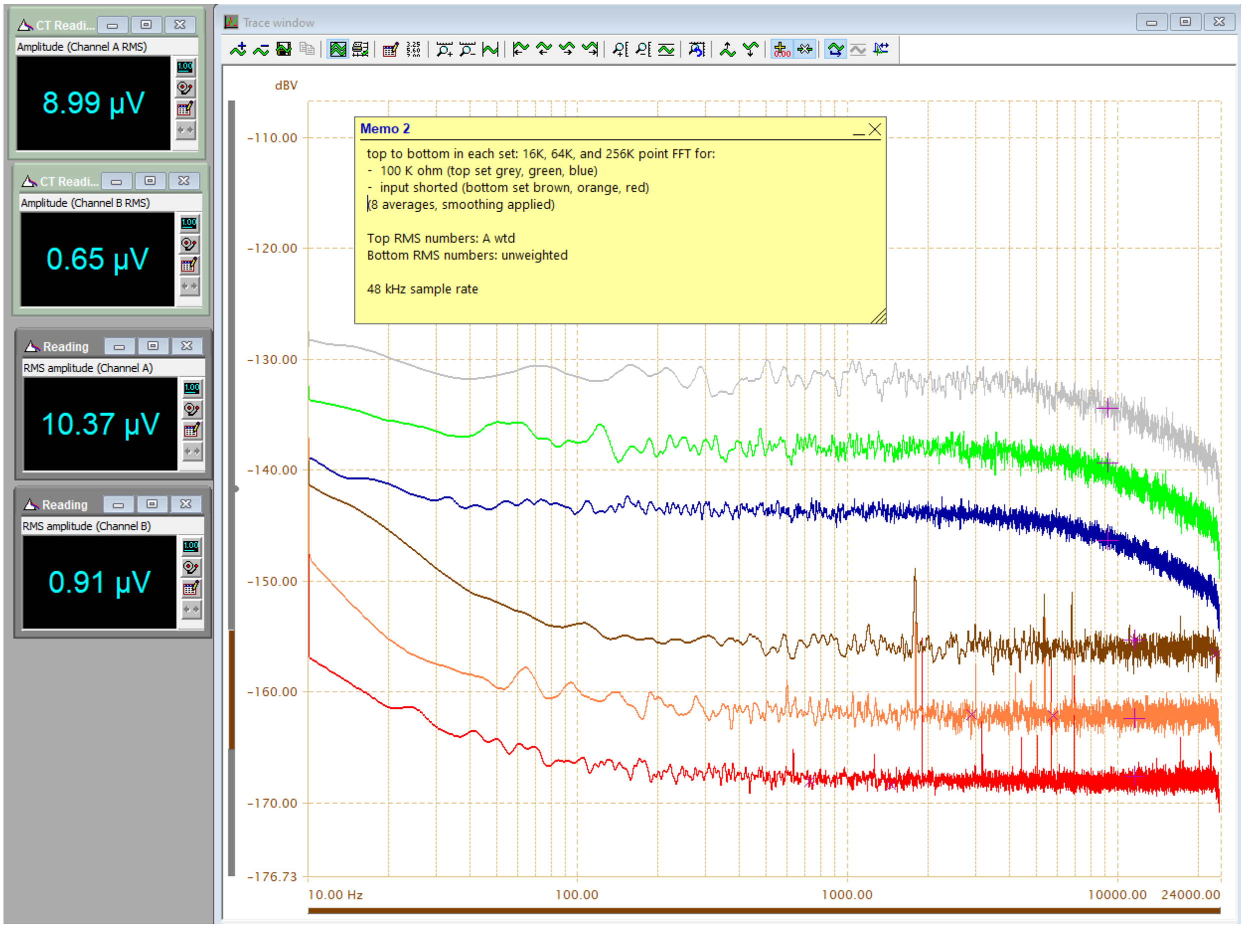

Figure 31 Example of FFT processing gain using the M1 with open input and shorted input and 3 different FFT sizes. The RMS measurements do not depend on the FFT results.

The HF rolloff in the open circuit case is assumed to be from the high impedance input (i.e., there's no source driving it) and whatever input capacitance is present in the system. The input capacitance is not specified for the M1.

The Nyquist/Johnson noise for a 100K ohm resistor is about 6 uVRMS in 24 kHz bandwidth, here we see the input noise increase from (unweighted) around 1 uV to 10 uV RMS, so there is something else happening in the input circuitry in the open (high impedance) case.

For a more direct comparison of the noise floor plotting Amplitude Spectral Density (ASD) or Power Spectral Density (PSD). ASD is the square root of PSD and is in units of $VHz^{-1/2}$ (volt-root-Hertz). While making noise comparisons easier, an ASD plot will change the plotted magnitude of tones because everything is normalized to a 1 Hertz equivalent bin width.

A.3 TONES AND AMPLITUDE FROM THE FFT

Though beyond the scope of this TechNote[9], changing the FFT size may have an impact on the harmonic distortion as computed from the measured peaks. If the frequency of the test tone for a THD test is synchronized with the sample rate and FFT size to have an integer number of cycles in the FFT then the amplitude measurement can be exact. If the test waveform does not have an integer number of cycles then the energy is spread across more than one FFT bin, the amounts of which depend on the window function used.

An analyzer like the dScope M1 will provide an estimate of the correction, however some small variation will be seen in a THD calculation as very low (harmonic) tones will also include some of the noise found in the bin(s) of the harmonic(s).

An example of this is seen comparing the data in Figure 22 (16K FFT) and Figure 23 (128K FFT) and copied here in to Figure 32 where the larger FFT resulted in the lower – by 1 dB – THD number. Looking at the individual components of the harmonics it's not quite as clear; some got smaller but some measured higher. The noise is random and the two figures are measurements taken at different points in time so there is a natural variation in the noise contribution to the FFT bin(s) with the harmonics.

[9] For an in-depth look at spectral analysis with the FFT see for example the open access book *"Window Functions and Their Applications in Signal Processing,"* by K.M.M. Prabhu, available from https://www.taylorfrancis.com/books/oa-mono/10.1201/9781315216386/window-functions-applications-signal-processing-prabhu

Marks : Ch A Live FFT Trace

Label	Frequency (Hz)	Amplitude (dBFS)
Fundamental	996.11	-3.20
Second	1992.23	-105.31
Third	2988.34	-122.58
Fourth	3987.39	-125.45
Fifth	4983.50	-114.94
Sixth	5979.62	-134.38
Seventh	6978.66	-127.85
Eighth	7974.78	-129.81
Sum of harmonics		-104.46 dBFS

Marks : Ch A Live FFT Trace

Label	Frequency (Hz)	Amplitude (dBFS)
Fundamental	996.85	-3.35
Second	1993.69	-106.29
Third	2990.91	-120.29
Fourth	3987.75	-127.56
Fifth	4984.60	-117.03
Sixth	5981.81	-140.50
Seventh	6978.66	-128.65
Eighth	7975.51	-134.84
Sum of harmonics		-105.57 dBFS

Figure 32 THD calculation from 16K and 128K FFT

Changing the FFT size also changes the bin spacing and harmonics may be proportioned differently between adjacent bins, which based on the window function being used and the way the actual peak is estimated may mean a slight change in the reported value as well.

The M1 supports a one million point FFT, and if we were patient enough to wait 3 minutes for eight averages, the result would have slightly better accuracy compared to the smaller FFT computed here. Given the chips may have performance difference of 6 dB or more and still meet the datasheet specifications measuring down to a fraction of a dB of computational accuracy just gives a false sense of righteousness to the measurement.

A.3.1 SMOOTHING AND PEAKS

Another way to make a mistake in computing levels is to enable smoothing. Smoothing the result can be very helpful when comparing two different FFT results and we want to see if the value across some range of frequencies changes in different tests. A typical FFT, even when averaged, might have *hair* of 5 dB or more in noise dominated areas and be difficult to visually average. Smoothing will reduce a large frequency peak (like a harmonic component) because the peak value is being averaged with presumably much lower noise values.

A.4 A-WEIGHTED MEASUREMENTS VS. UNWEIGHTED MEASUREMENTS

Looking further at Figure 31 we can see the A-weighted measurements have smaller value than unweighted measurements, which is expected because the A-weighting filter removes low and high frequency to match the human ear's sensitivity at low levels[10], thus making the value more predictive in terms of audibility of the noise.

A-weighting reduces the RMS value by about 1.4 dB for the open input (which has 100 k ohm impedance) and the shorted input by 3 dB. This difference is because neither the open input nor the shorted input examples are spectrally flat, so the A-weighting curve has a different apparent effect on the noise measurements.

As already discussed, the single noise level number does not give a complete picture compared to examining the spectrum. Barring very large amplitude tones in the noise spectrum, noise level number does provide a reasonable estimate for how audible noise might be in an audio system.

[10] At high sound levels the ear's sensitivity flattens and A-weighting is not appropriate.

APPENDIX B SIGMA STUDIO SETTINGS

Audio setup for AD2418 to match I^2S setting used by AKM EVM for I^2S mode in the AKM EVM software (left-most choice in the AKM PC software).

Figure 33 AD2428 I^2S settings for connection to the AKD4619 EVM