P1658 Digital to Analog Standard Meeting, May 19, 2008

Attendees:
Robert Graham, Sandia National Labs., Albuquerque, NM USA rmgraha@sandia.gov
Richard Liggiero, LTX Corporation, Norwood, MA USA* Richard_Liggiero@ltx.com
Tom Linnenbrink, Hittite, (Chair, TC-10) linnenbrink@hittite.com
Solomon Max, LTX Corporation, Norwood, MA USA (Secretary) sol_max@ltx.com
Pasquale Daponte, University of Sannio, Benevento, Italy (Editor) daponte@unisannio.it
Bill Haderig, Tektronix Corporation
Jerry Blair, National Security Technologies LLC, Las Vegas, NV USA blairjj@earthlink.net

*Telephone-WebEx

Agenda: (approximate)

1. Introductions of any new members, if needed
2. Review and approve the minutes of the last meeting.
3. Review of status of assignments per minutes of previous meetings
4. Review of status of the draft
5. Other business, PAR extension, etc.
6. Detailed review of draft
7. Break for lunch
8. Detailed review of draft
9. Review action items
10. Adjourn and select dinner location

Introductions were made for all attendees.

The minutes from the previous meeting (15-November-2007) were read and accepted.

The following is a list of assignments that were made at the November meeting. The committee reviewed the progress in the assignments.

#	Assigned Checker	Section	Assigned Reviewer	Comment
1	Solomon Max	1.2	Michaeli	Digital to Analog Converter Background
2	Steve Tilden	1.4.3	Solomon Max	Additional Specifications
3	Fang Xu	3.1.19	Solomon Max	dynamic range
4	Solomon Max	3.1.x	Fang Xu	differential gain/ differential phase
5	Haasz	3.1.28	Solomon Max[2]	Harmonic
6	Niclas	3.1.43	Fang Xu[3]	Noise Power Ratio
7	Haasz	3.1.52	Fang Xu	Pass band
8	Fang Xu	3.1.83 3.1.84	Solomon Max Steve Tilden	THD THD+N

Assignment #1 was completed by Solomon Max. The checking process has not yet been completed by Professor Michaeli. Jerry Blair reviewed the section and said that it was acceptable. It was added to the draft.

Assignment #2 was not completed by Steve Tilden.

Assignment #3 currently reads as follows;

dynamic range: This term has a number of meanings depending on the application. It is not used by itself in this standard. See: the specific related parameters: spurious free dynamic range, effective number of bits, signals to noise and distortion ratio, and signal to noise ratio**.** The ratio of the maximum output level of a DAC, to the minimum detectable DAC output change.

Dr. Fang Xu will try to improve on the current text. Some areas of concern relate to the fact that by loading the DAC at a very fast rate, it is possible to synthesize signals arbitrarily smaller than 1lsb. It might be appropriate to specify the dynamic range as the ratio of the rms of the largest signal that the DAC is able to generate (FSR/2/$\sqrt{2}$) to the rms noise over a specified bandwidth. (Comment by Solomon Max).

Assignment #4 was reviewed by the committee. It was recommended that it be split into two parts. Solomon Max will rewrite them. The current text in the draft now reads;

3.1.68 differential gain and phase: These tests that were historically performed for video and television applications where a chroma subcarrier was used. Differential gain is defined as the percentage difference between the output amplitudes of a small chroma sinewave superimposed on two defined voltage levels. Differential phase is defined as the difference in the output phase of a small chroma sinewave superimposed on two defined voltage levels.

3.1.68 differential gain: The ratio of the maximum gain and the minimum gain of a small-amplitude sinewave that is superimposed on multiple static levels, divided by the average gain.

3.1.68 differential phase: The difference in the output phase of a small high frequency sinewave for two stated levels of a low frequency signal on which it is superimposed.

The other assignments were not completed

Jerry Blair has agreed to write a section on interpolation filters. The discussion on bandwidth will be considered an ongoing discussion

The section on NPR (Noise Power Ratio) will be left incomplete until a test method section is generated.

THD and IMD were discussed. The Nyquist limit was removed. A note was added – "No final decision has been made concerning the fold-back harmonics. They may or may not be included as part of THD". The comment is necessary for the following definitions;

3.1.84 total harmonic distortion (THD): The ratio of the square root of the sum of the squares of the rms amplitudes of the harmonic components of the output sinewave (i.e. of sinusoidal components appearing at multiples of output sinewave frequency) to the rms value of the output sinewave. It is typically expressed in decibels.

3.1.85 total harmonic distortion plus noise (THD+N): The ratio of the square root of the sum of the squares of the amplitudes of all components of the output signal (excluding the fundamental) to the amplitude of the fundamental output, over a specified frequency band. It is typically expressed in decibels. It is similar to THD, but includes the noise.

3.1.89 two tone intermodulation distortion (IMD): The ratio of the rms amplitude of either of two equal-amplitude input tones to the rms amplitude of the worst third order intermodulation product; reported in dBc.

Pasquale Daponte is asked to change the term "value" to "amplitude" as appropriate. An example would be

1.1.1 Digital-to-analog converter and analog-to-digital converter differences and similarities

Digital to Analog Converters (DACs) and Analog to Digital Converters (ADCs) perform complementary functions. DACs receive a digital signal and generate an analog equivalent, while ADCs receive an analog signal and generate a digital equivalent. Some of the subtleties are summarized below.

1) Transfer function differences (See Figure 2)
   a) For a DAC, each digital input corresponds to a single average analog output value.
   b) For an ADC, each digital output corresponds to an interval of analog input values. The analog quantity defined in the transfer function of an ADC, the code edge, is defined to occur at the point where 50% of the output codes are greater than a specified digital output

Should be replaced by;

1.1.2 Digital-to-analog converter and analog-to-digital converter differences and similarities

Digital to Analog Converters (DACs) and Analog to Digital Converters (ADCs) perform complementary functions. DACs receive a digital signal and generate an analog equivalent, while ADCs receive an analog signal and generate a digital equivalent. Some of the subtleties are summarized below.

2) Transfer function differences (See Figure 2)
   a) For a DAC, each digital input corresponds to a single average analog output amplitude.
   b) For an ADC, each digital output corresponds to an interval of analog input values. The analog quantity defined in the transfer function of an ADC, the code edge, is defined to occur at the point where 50% of the output codes are greater than a specified digital output

Section 3.1.92 currently reads as follows;

3.1.92 update rate (f_u): The rate at which the digital signal is converted to the analog signals are generated. Analog output values are changed by the DAC input
The section needs some thought. Jerry Blair will review it and attempt to come up with a better definition that considers things like digital filtering, and DACs with internal digital dividers.

Professor Michaeli inserted a comment.

"I am missing parameters important for "bit stream DAC" like: Ripple voltage, Number of bits M at the output of DAC, over sampling ratio (OSR) etc. For other DACs Thermal drift, SNDR – see Plassche.")"

The meaning of the phrase "Number of bits M at the output of DAC" was unclear to those at the meeting. Professor Michaeli is asked to comment,

Section 3.1.68 reads as follows;

3.1.68 resolution: Number of bits representing an analog signal. The higher the number of bits, the higher the resolution of the converter. Generally more accurate too.

Solomon Max will work on improving the definition to include some of the items referred to by Professor Michaeli.

The expression U[k] will be used as the output amplitude of a DAC for an input code k. Note that the input code may not correspond to the input data value. K is a number that increases monotonically from a value that corresponds to negative full scale amplitude to a value that corresponds to a positive full scale value.

The definition of sigma was modified as follows;

σ = standard deviation; sometimes used as a measure of noise amplitude, which is the standard deviation of the random component of a signal.

f_u (Update Frequency) has now replaced f_s (Sample frequency)

It was noted that the symbols M_{test}, N_{test}, $M_{digitize}$, and $N_{digitize}$ are used in the section related to test setup in Figure 8. It is not clear that they belong as standard symbol definitions.

It was decided that INL and DNL belonged both as abbreviations and symbols.

Section 4.1.1 (Test setup) will be edited by Solomon Max.

Professor Pasquale has agreed to develop a standard method for dealing with measurement uncertainties. It was unclear as to whether the uncertainties should be evaluated in the standards, or left to the user (since many of the uncertainties are instrument-specific)
It was proposed that the section numbers be changed. The following section numbering scheme was proposed. The section numbering should be organized so as to make the Table of Contents more useful.

4.0 General
5.0 Test setup
6.0 Taking a record of data
7.0 Test method – Static transfer characteristics
8.0 Test method – Dynamic characteristics

.

.Annex A Fitting sinewaves

Annex B DFT

.

.In section 4.1.2.4 the following text was observed;

The sinewave curve fit is used in several specific test methods described later in this document. Analyze the residuals and fit parameters using methods described for the specific test being performed.

The section needs rewording. Solomon Max will try rewording it.

Equation 10 reads as follows;

For any value of *J*, relatively prime to *M*, there is a unique value, I, between 0 and *M* -1, which satisfies *IJ* = 1 Mod *M*. The number *I* is the inverse (Mod *M*) of *J*, and its value determines the frequency accuracy required.

For an exact input frequency the maximum difference between successive phases is $2\pi/M$. *If the frequency does not have the exact value specified in Equation (9) the maximum phase difference will be larger.* If the larger value is written in the form $(1+\rho) \times (2\pi/M)$, the error, ε_f, in the frequency must satisfy

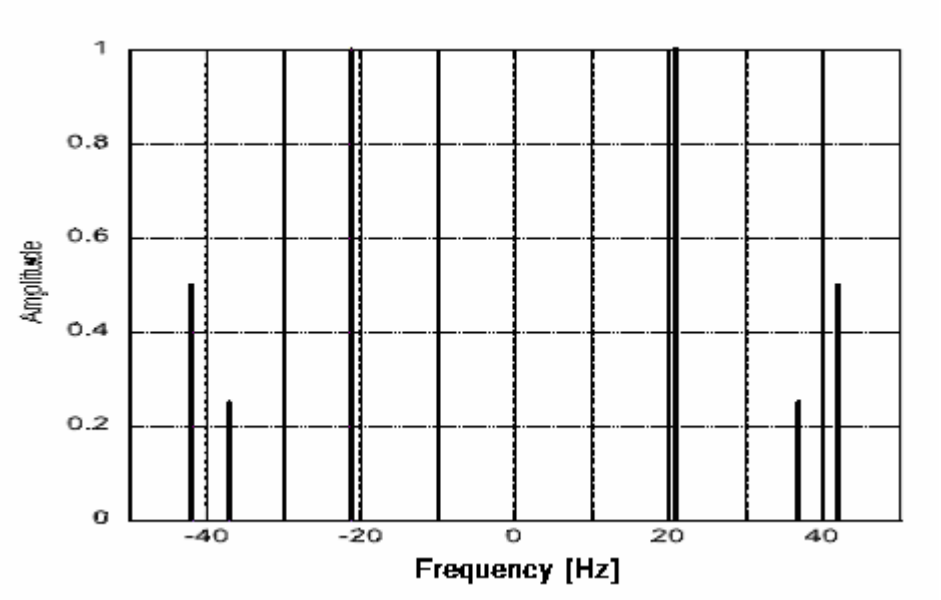
$$\begin{aligned} \left|\frac{\varepsilon_f}{f_i}\right| &\le \frac{\rho f_i}{IMf_i} && for\ \varepsilon_f > 0 \\ \left|\frac{\varepsilon_f}{f_i}\right| &\le \frac{\rho f_i}{(M-I)IMf_i} && for\ \varepsilon_f < 0 \end{aligned} \qquad (10)$$

where

ε_f	is the error in input frequency
ρ	is defined above
f_i	is the input frequency
I	is the inverse (Mod M) of J
M	is the record length.

Jerry Blair will investigate Professor Haasz's concerns.

Figure 10 was modified to explicitly use the units Hz for the horizontal axis. See modified Figure 10 below;



It was recommended that the sections on DFT be moved to an Annex.

Dr. Sedlacek from the University of Sannio had some recommendations concerning windowing. The committee asked him to write up his suggestions for implementing his suggestions.

Section 4.2.3 includes some discussions about a zero order hold. This should be included in a different section near the beginning of the standard. (Section 1.2.?) Jerry Blair will work on moving the section into 1.2

Static input parameters are reassigned to Richard Liggiero. (Section 4.2.3)

The t_s in Figure 15 should be changed to t_{su}. The changes will be implemented by Richard Liggiero.

Solomon Max will rewrite the section on dynamic gain error. (section 4.5.3) A discussion will be included on methods of separating the frequency dependent gain from the reconstruction filter characteristics.

The section on SFDR test method (Section 4.6.4) was not written in the standard format. The standard format should include a brief description of the parameter being tested in a first section (4.2.3). A second section (4.2.3.1 should give the test method. If necessary a third section (4.2.3.2) should include a discussion of the test.

A new Figure 17 is required with more information concerning what is displayed. The current Figure 17 is shown below.

Figure 17 illustrates the block diagram of a circuit configuration which can measure SINAD, SNR, and THD. A plot of the resultant spectrum is shown in the figure.

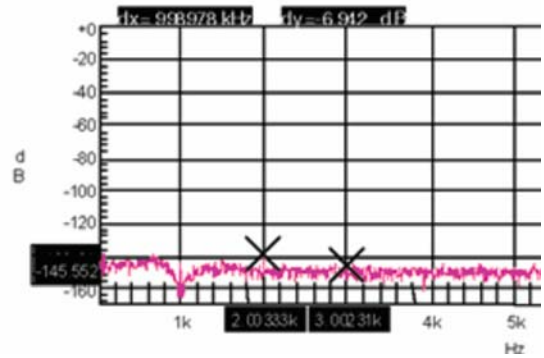


Figure 17 –

The repair of the section is assigned to Solomon Max

The following is a table of assigned tasks. Those assigned are asked to complete their tasks before September of 2008 so that they can be reviewed at IMEKO in Florence.

#	Assigned Checker	Section	Assigned Reviewer	Comment
1	Fang Xu	3.1.19	Solomon Max	dynamic range
2	Solomon Max	3.1.x	Fang Xu	differential gain/ differential phase
3	Jerry Blair	1.2	Solomon Max	Interpolation Filters
4	Pasquale Daponte		Solomon Max	Change "value" to "amplitude"
5	Solomon Max	3.1.68	Michaeli	Resolution definition
6	Solomon Max	4.1.1	Pasquale Daponte	Test Setup
7	Pasquale Daponte		Solomon Max	Measurement Uncertainties
8	Pasquale Daponte		Steve Tilden	Reorganize Section Numbering
9	Solomon Max	4.1.2.4	Jerry Blair	Sine fit rewording
10	Jerry Blair	Eq. 10	Haasz	Point re-ordering
11	Sedlacek		Jerry Blair	Windowing
12	Richard Liggiero	4.2.3	Solomon Max	Static input parameters
13	Solomon Max	4.5.3	Jerry Blair	Dynamic Gain Error
14	Solomon Max	4.6.4	Steve Tilden	SFDR test method
15				

The Assignments should be implemented by going through the following steps

1. Copy the specified sections into a Word document named "Sectionx.x.x.mod.doc", accept all changes. (x.x.x refers to the specific section number)
2. Set word into the mode of Track Changes, Highlight Changes.
3. Edit the document carefully to correct technical errors, or clarify the wording.
4. Send the document to the reviewer.
5. Implement this before August 15, 2008.

The reviewer should proceed through the following steps.

1. Review the document
2. Edit the document in the mode of Track Changes, Highlight Changes.
3. Send the reviewed document to sol_max@ltx.com and to Pasquale Daponte (daponte@unisannio.it) with the filename "Sectionx.x.x.mod1.doc" (x.x.x refers to the specific section number). The secretary will post the document on the web site. Professor Daponte will incorporate the modifications into the newest draft. Complete this by September 1, 2008.

Solomon Max June 19, 2008