Measurements of Various Intermodulation Distortions (IMD, TD+N, DIM) using Multi-Instrument

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This article presents how to measure various Intermodulation Distortions (IMD, DIM, TD+N) correctly using Multi-Instrument. Sophisticated mathematics is intentionally avoided in this article in order to make it easily understood by most software users.

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1. Introduction

1.1 Distortion Classification

Simply put, distortion is the alteration of the waveform of a signal when it passes through a system. Distortions can be classified as either linear or non-linear distortions.

1.1.1 Linear Distortion

A linear distortion is a distortion with no new frequencies added. It can be caused by the non-flat magnitude frequency response or the non-linear phase frequency response of the system.

1.1.2 Non-linear Distortion

A non-linear distortion is a distortion with new frequencies added. It can be classified as either harmonic or non-harmonic. Non-linear distortions are sometimes simply called “distortions”.

There are other ways to describe non-linear distortion, such as non-coherent distortion and GedLee Metric, etc.

1.1.2.1 Harmonic Distortion

Harmonic distortions refer to those newly added frequencies which are integer multiples of the fundamental frequency.

For more information, please refer to the article “Measurement of Total Harmonic Distortion and Its Related Parameters Using Multi-Instrument” at https://www.virtins.com/doc/Measurement-of-Total-Harmonic-Distortion-and-Its-Related-Parameters-using-Multi-Instrument.pdf.

1.1.2.2 Non-Harmonic Distortion

Non-harmonic distortions refer to those newly added frequencies which are not integer multiples of the fundamental frequency, such as intermodulation distortion. Note that noises are not considered as non-harmonic distortions.

1.2 Overview

This article describes the measurements of non-harmonic distortions, in particular how to measure intermodulation distortions. Unlike the harmonic distortion measurement which uses a single-frequency sine wave as the stimulus, the non-harmonic distortion measurement uses the sum of two or more sine waves of different frequencies as the test signal. When a two-tone or multi-tone test signal is fed into a nonlinear system, the intermodulation between the frequency components forms additional frequencies which are not harmonically rated to any of the stimulus frequencies. These new frequency components are referred to as intermodulation distortions. Various intermodulation distortion measurement techniques exist. Two-Tone Intermodulation Distortion (IMD, such as SMPTE/DIN IMD, CCIF2 IMD, CCIF3 IMD), Multi-Tone Total Distortion Plus Noise (TD+N), and Dynamic Intermodulation Distortion (DIM) will be introduced in this article.

To measure intermodulation distortion, a signal generator is employed to generate the test signal with sufficiently low distortions. The generated signal is used as the stimulus to the Device Under Test (DUT). Meanwhile, the response from the DUT is sampled and then analyzed using Fast Fourier Transform (FFT). The signal power is decomposed into four parts: fundamentals, harmonics, intermodulation frequencies, and noise. The DC component is usually filtered out and not used in the calculation. Finally, an overall intermodulation distortion value is calculated based on its definition formula.

Similar to THD measurement, software loopback tests can be used to verify the correctness of the test parameters in order to avoid FFT artefacts and the concentration of quantization noise spectrum. The performance of the measuring device can be verified through hardware loopback tests. It must be substantially better than that of the DUT in order to ensure measurement accuracy. For more information, please refer to the article “Measurement of Total Harmonic Distortion and Its Related Parameters Using Multi-Instrument” at https://www.virtins.com/doc/Measurement-of-Total-Harmonic-Distortion-and-Its-Related-Parameters-using-Multi-Instrument.pdf.

2 Two-tone Intermodulation Distortion (IMD)

The stimulus used in a two-tone intermodulation distortion measurement consists of two frequency components which are generally not harmonically related. The commonly used two-tone intermodulation distortion measurement techniques are SMPTE / DIN IMD, CCIF2 IMD and CCIF3 IMD. They are different in the stimulus's frequency components and the intermodulation products used for evaluation. In Multi-Instrument, these measurement modes can be selected by right clicking anywhere within the Spectrum Analyzer window and selecting [Spectrum Analyzer Processing]> "Parameter Measurement"> "IMD".

2.1 Two-tone Intermodulation Products

When a two-tone signal passes through a nonlinear system, an infinite number of intermodulation frequencies of the two tones will be produced. They can be expressed as:

$$mf_L \pm nf_H$$

where f_L and f_H are the frequencies of the two tones ($f_L<f_H$), m and n are positive integers. The value of m+n represents the order of the intermodulation term. Similar to the case of harmonic production, odd-order intermodulation products are caused by symmetrical non-linearity in the device transfer function and even-order intermodulation products are caused by non-symmetrical non-linearity.

Fig.1 shows the measured output signal spectrum of a system excited by a linear sum of 60 Hz and 7000 Hz with an amplitude ratio of 4:1. The output signal contains the 60 Hz and 7000 Hz fundamentals, harmonics, intermodulation products and noise. When f_L<<f_H, most of $mf_L \pm nf_H$ appear as sidebands around nf_H.

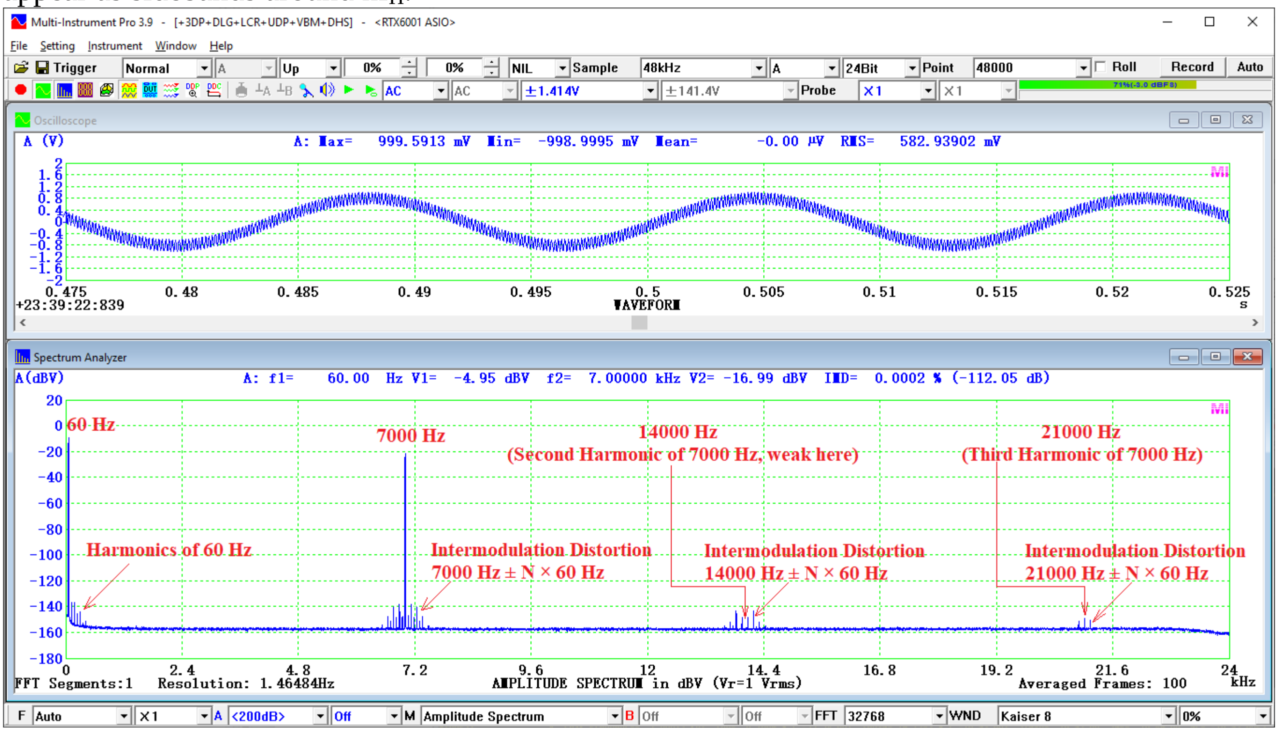

Fig. 1 The output signal spectrum of a system under the stimulation of 60 Hz and 7000 Hz with an amplitude ratio 4: 1

2.2 Composite Frequency (or Waveform Repetition Rate) of a Two-tone Signal

The concentration or uneven distribution of quantization noise spectrum is caused by its correlation with the signal frequency, or the composite signal frequency (i.e. the waveform repetition rate) if the signal is not sinusoidal. If the two-tone signal is periodic, then its composite frequency f satisfies the following equation:

$$\frac{1}{f} = \frac{N_L}{f_L} = \frac{N_H}{f_H}$$

That is, one cycle of the signal contains N_L cycles of f_L and N_H cycles of f_H, where N_L and N_H are coprime (i.e. the greatest common factor of them is 1) and satisfies:

$$\frac{N_L}{N_H} = \frac{f_L}{f_H}$$

For example, in a SMPTE IMD test signal, f_L = 60 Hz, f_H = 7000 Hz, then:

$$\frac{N_L}{N_H} = \frac{f_L}{f_H} = \frac{60}{7000} = \frac{3}{350}$$

i.e.: N_L = 3 and N_H=350, thus f = f_L / N_L = 60 / 3 = 20 Hz.

Similarly, we have:
DIN IMD: f_L = 250 Hz, f_H = 8000 Hz, N_L=1， N_H=32， f = 250 Hz
CCIF2 IMD: f_L=19000 Hz, f_H = 20000 Hz, N_L = 19， N_H = 20， f = 1000 Hz
CCIF3 IMD: f_L=13000 Hz, f_H = 14000 Hz, N_L = 13， N_H = 14， f = 1000 Hz

2.3 How to Avoid or Suppress Spectral Leakage

Similar to THD measurement, IMD measurement is also sensitive to spectral leakage. To avoid spectral leakage, a FFT segment must contain an integer number of signal cycles. This can be expressed in math as:

[Sampling Rate] / [Signal Frequency] = [FFT Size] / [Number of Cycles]

Sampling that satisfied the above condition is called full-cycle sampling. FFT size here must be a power of 2. The composite frequencies of SMPTE IMD, DIN IMD, CCIF2 IMD and CCIF3 IMD are 20 Hz, 250 Hz, 1000 Hz and 1000 Hz, respectively. Given the commonly used sampling rates: 44.1 kHz, 48 kHz, 50 kHz, 96 kHz, 100 kHz, 192 kHz, 200 kHz, etc., it is almost impossible to use full-cycle sampling here. Therefore, spectral leakage is inevitable and a window function must be used to suppress it. For IMD measurement, window functions that are able to confine most of the energy of a periodic component in its neighboring FFT bins are preferred. These window functions usually have a big main lobe. Kaiser 6 ~ Kaiser 20, Blackman Harris 7, Cosine Sum 220, Cosine Sum 233, Cosine Sum 246, Cosine Sum 261 are recommended. More information on various window functions can be found at:

https://www.virtins.com/doc/D1003/Evaluation_of_Various_Window_Functions_using_Multi-Instrument_D1003.pdf

2.4 How to Avoid Quantization Noise Being Measured As IMD

When sampling a periodic signal f, the quantization noise is always concentrated at the fundamental and harmonics of a frequency equal to the greatest common factor f_{gcf} of the sampling rate f_s and signal frequency f. The smaller the f_{gcf}, the more randomized the quantization noise. When the f_{gcf} is big, slightly changing the two tone frequencies to make them less reducible or even irreducible to each other will minimize their composite frequency f and in turn have a better chance to minimize the f_{gcf}. The effect of the quantization noise on IMD measurement will be discussed later in each type of IMD tests.

2.5 SMPTE / DIN IMD (or MOD IMD)

SMPTE/DIN IMD is the most common IMD measurement. SMPTE (Society of Motion Picture and Television Engineers) standard RP120-1983 and DIN (Deutsches Institut für Normung) standard 45403 are similar. Both specify a two-tone test signal consisting of a large amplitude low-frequency tone linearly mixed with a high-frequency tone at ¼ of the amplitude of the low frequency tone. SMPTE specifies 60 Hz and 7 kHz mixed at 4:1. The DIN specification allows several choices in both frequencies, with 250 Hz and 8 kHz being the most common. The IMD under this category is defined as the square root of the ratio of the power of the sidebands to the power of the upper frequency. It can be expressed in percentage (%) or dB. Traditionally, SMPTE / DIN IMD is measured using an analog amplitude demodulation technique. The measured IMD includes the contribution of noise within the pass band. With FFT, individual intermodulation products can be readily measured. This is the method adopted by Modulation IMD (MOD IMD) to reduce the influence of noise. The sidebands used in MOD IMD calculation are f_H-f_L, f_H+f_L, f_H-2f_L, f_H+2f_L, where f_H and f_L are the high frequency and low frequency of the two tones respectively. The following formula shows how MOD IMD is calculated in Multi-Instrument.

$$\text{MOD IMD} = \frac{\sqrt{(V_{f_H-f_L} + V_{f_H+f_L})^2 + (V_{f_H-2f_L} + V_{f_H+2f_L})^2}}{V_{f_H}} \times 100\%$$

where V_x is the RMS amplitude of the frequency component *x*.

IMD can also be expressed in dB:

$$(\text{MOD IMD})_{dB} = 20\log_{10}(\text{MOD IMD})$$

For example, if MOD IMD = 0.0001%, then $(\text{MOD IMD})_{dB}$ = -120 dB.

2.5.1 SMPTE IMD Software and Hardware Loopback Tests

The SMPTE IMD test signal can be generated using the MultiTone function of the Signal Generator in Multi-Instrument. Its configuration is shown in Fig. 2.

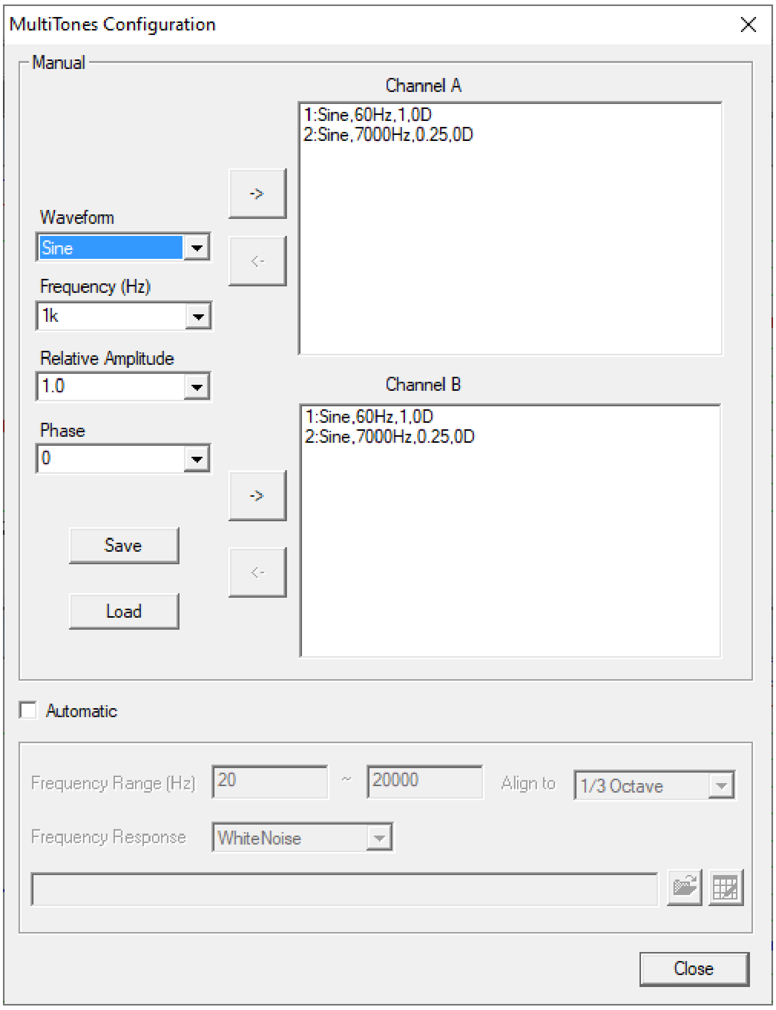

Fig. 2 SMPTE IMD Test Signal Configuration

Fig. 3 shows the 24-bit software loopback test results of SMPTE IMD. The ideal test signal is generated using the "iA=oA, iB=oB" software loopback mode of the Signal Generator of Multi-Instrument. The test parameters are: [Sampling Rate] = 48 kHz, [Frequency Composition] = 60 Hz + 7000 Hz (with an amplitude ratio 4:1), [FFT Size] = 32768, [Sampling Bit Resolution] = 24, [Window Function] = Kaiser 8. No zero padding is applied as [Record Length] = 48000 which is greater than the FFT size (Note: Multi-Instrument is able to measure IMD correctly even with zero padding). The sidebands used in SMPTE IMD calculation are 6880Hz, 6940Hz, 7060Hz and 7120Hz. The measured SMPTE IMD is 0.0000100% (-140.03 dB). As shown previously, the composite frequency of the standard SMPTE IMD test signal is 20 Hz. The greatest common factor of 20 Hz and the sampling rate 48000 Hz is also 20 Hz. Hence, the quantization noise is concentrated at 20 Hz and its harmonic frequencies. The 20 Hz frequency interval is quite small and thus the quantization noise is distributed quite evenly along the frequency axis. Only a very little portion of it falls into the sidebands used in the SMPTE IMD calculation. This test shows that the residual SMPTE IMD due to the software is negligibly small for any practical SMPTE IMD measurements.

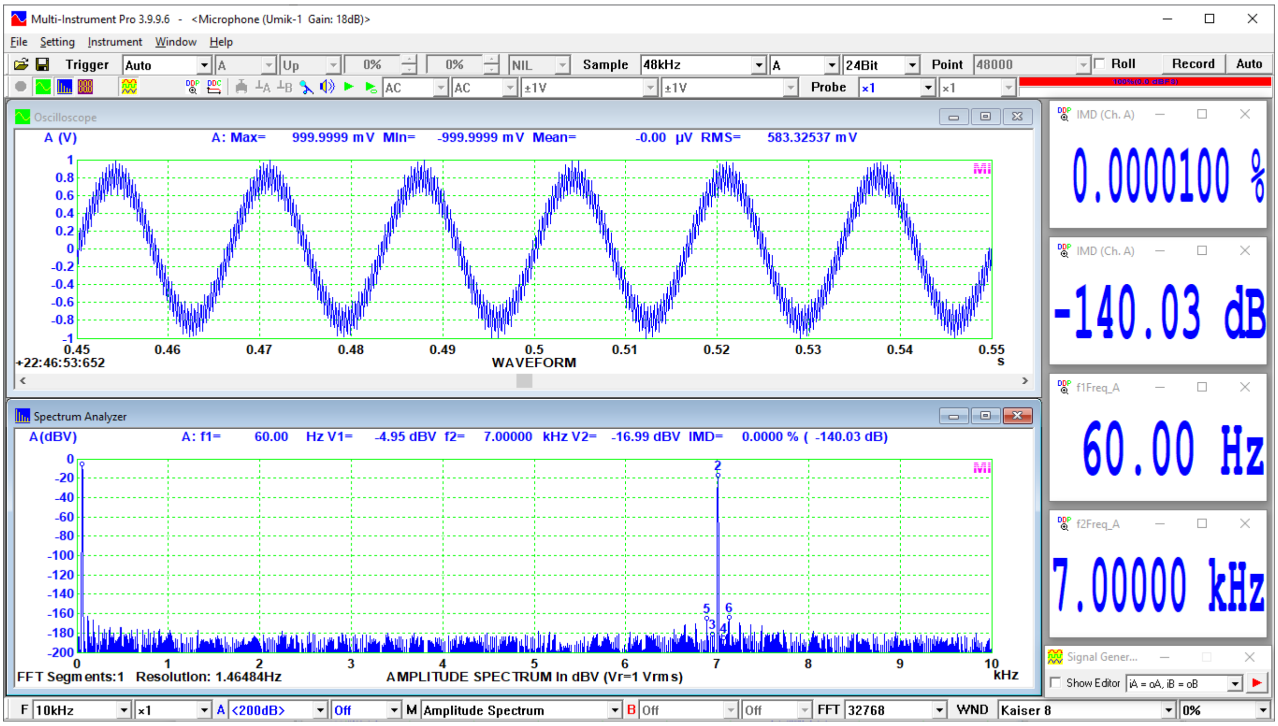

Fig. 3 SMPTE IMD Software Loopback Test

Fig. 4 shows the 24-bit hardware loopback test results of SMPTE IMD of a RTX6001 audio analyzer. The measured SMPTE IMD is 0.0002239% (-113.00 dB).

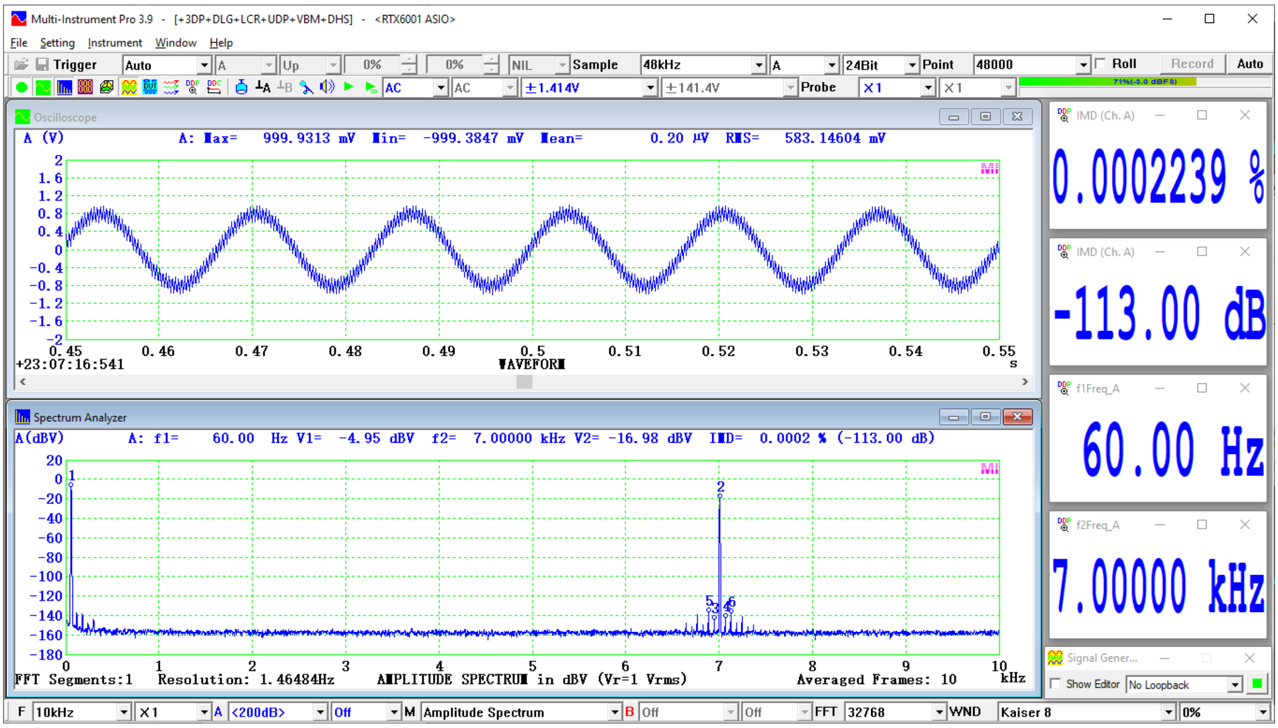

Fig. 4 SMPTE IMD Hardware Loopback Test (RTX6001)

2.5.2 DIN IMD Software and Hardware Loopback Tests

The DIN IMD test signal can be generated using the MultiTone function of the Signal Generator in Multi-Instrument. Its configuration is shown in Fig. 5.

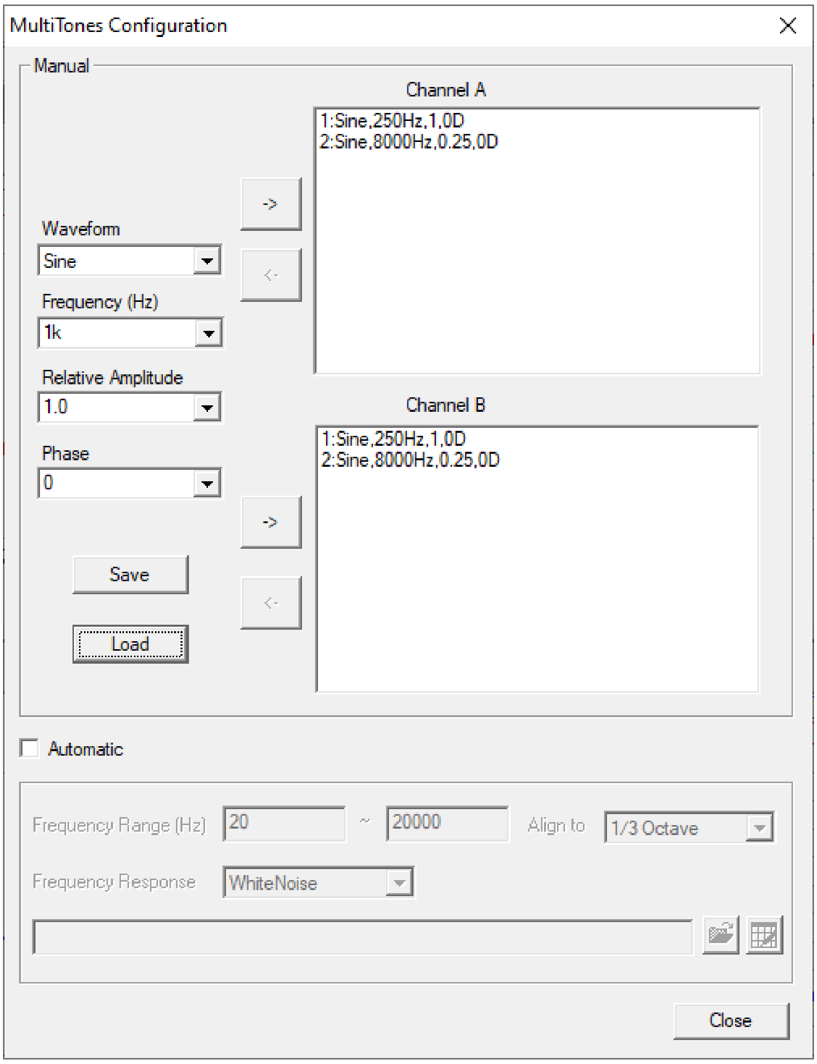

Fig. 5 DIN IMD Test Signal Configuration

Fig. 6 shows the 24-bit software loopback test results of DIN IMD. The ideal test signal is generated using the "iA=oA, iB=oB" software loopback mode of the Signal Generator of Multi-Instrument. The test parameters are: [Sampling Rate] = 48 kHz, [Frequency Composition] = 250 Hz + 8000 Hz (with an amplitude ratio 4:1), [FFT Size] = 32768, [Sampling Bit Resolution] = 24, [Window Function] = Kaiser 8. No zero padding is applied as [Record Length] = 48000 which is greater than the FFT size. The sidebands used in DIN IMD calculation are 7500Hz, 7750Hz, 8250Hz and 8500Hz. The measured DIN IMD is 0.0000105% (-139.59 dB). As shown previously, the composite frequency of the standard DIN IMD test signal is 250 Hz. The greatest common factor of 250 Hz and the sampling rate 48000 Hz is also 250 Hz. Hence, the quantization noise is concentrated at 250 Hz and its harmonic frequencies. The 250 Hz frequency interval is not very small and thus discrete frequency peaks at an interval of 250 Hz can be clearly seen along the frequency axis. A small portion of it falls into the sidebands used in the DIN IMD calculation. Nevertheless, this test shows that the residual DIN IMD due to the software is small enough for any practical DIN IMD measurements. Furthermore, the noise in the hardware will help to randomize the 24-bit quantization noise, as shown in Fig. 7.

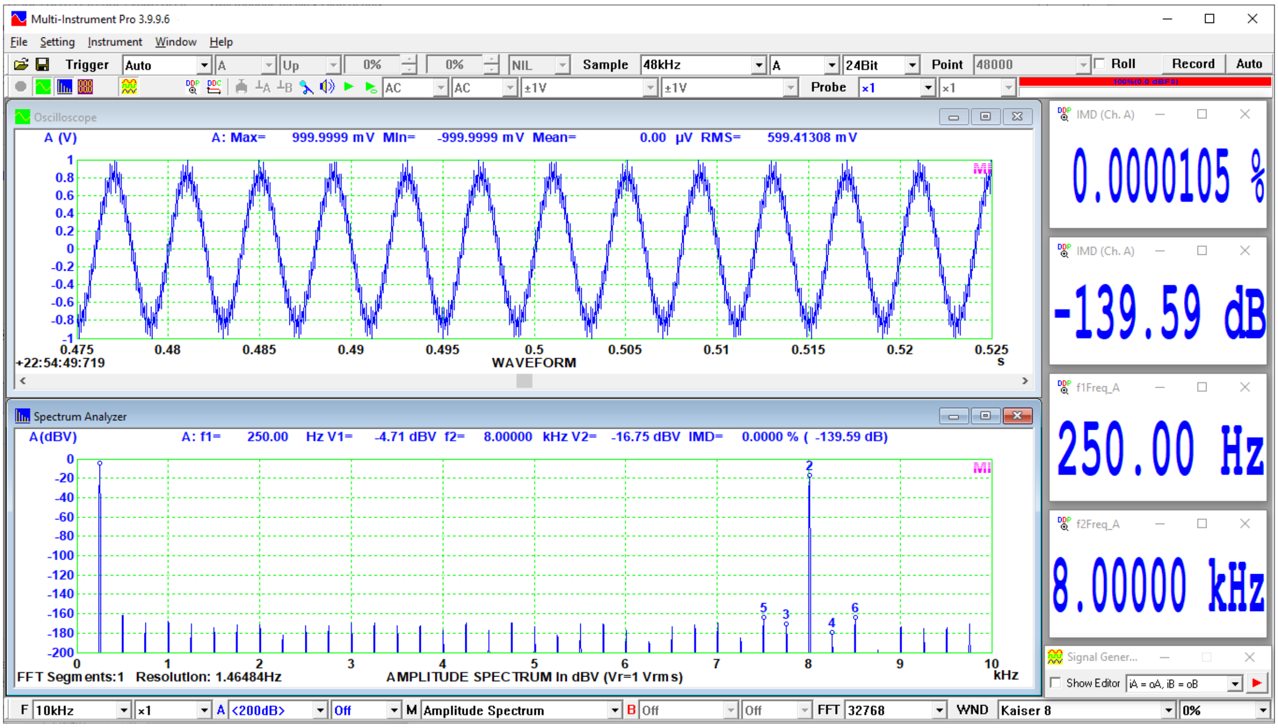

Fig. 6 DIN IMD Software Loopback Test

Fig. 7 shows the 24-bit hardware loopback test results of DIN IMD of a RTX6001 audio analyzer. The measured DIN IMD is 0.0002164% (-113.29 dB).

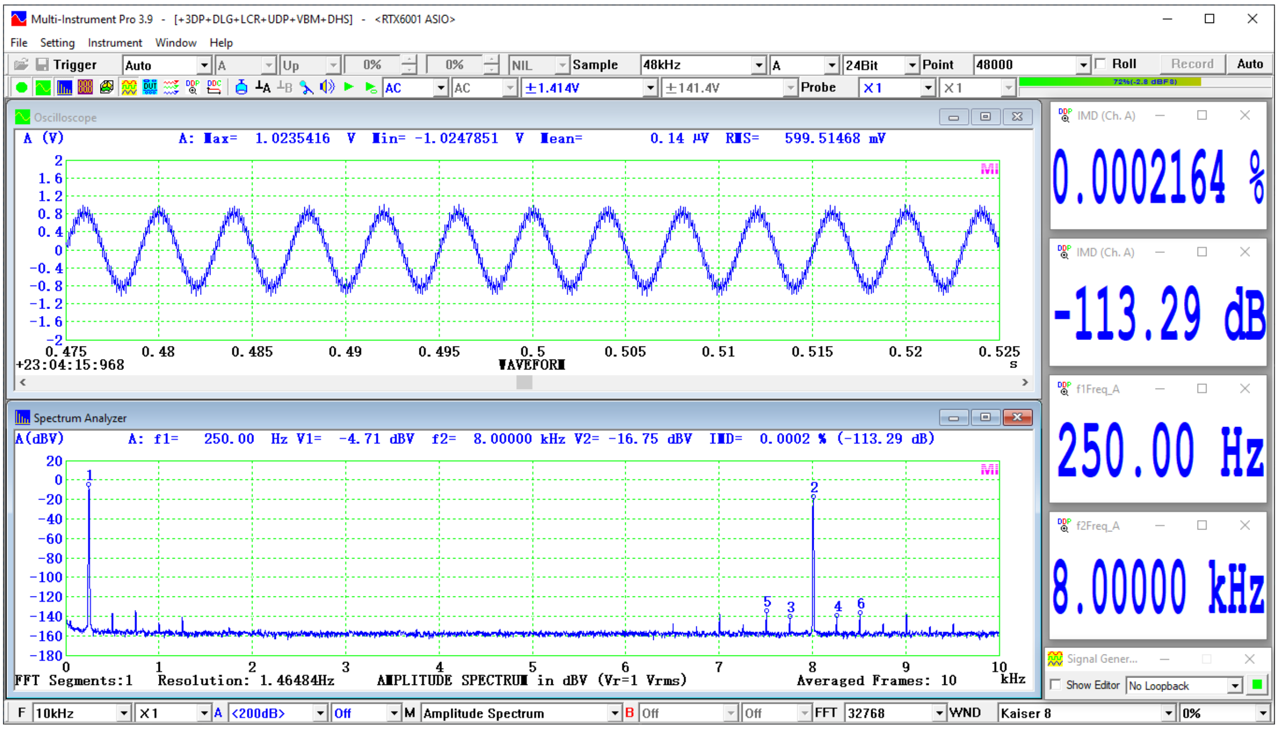

Fig. 7 DIN IMD Hardware Loopback Test (RTX6001)

2.6 CCIF IMD (or DFD IMD)

CCIF IMD, also called Twin-Tone IMD, Difference-Tone IMD, or Difference Frequency Distortion (DFD), is another common IMD measurement. It is described in IEC60118 and

IEC60268. The test specifies two equal-amplitude closely spaced high frequency signals. The IMD under this category is defined as the square root of the ratio of the power of the intermodulation distortion products to the square of the RMS amplitude sum of the two test frequencies. It can be expressed in percentage (%) or dB. It has two sub-types: CCIF2 IMD and CCIF3 IMD. Because the twin tones are high in frequency and their composite waveform has a steep slope, CCIF IMD is useful for observing nonlinear distortion that increases with frequency, such as slope induced distortion. As much of the distortion energy is contained in the intermodulation frequencies below or near the twin tones, CCIF IMD is a good choice for measuring nonlinear distortion in bandlimited devices where harmonic distortion products from a high-frequency stimulus would fall out of band.

2.6.1 CCIF2 IMD Software and Hardware Loopback Tests

For CCIF2 IMD, the commonly used twin tones are: 19 kHz and 20 kHz. The intermodulation distortion product(s) used for this type of IMD calculation is: f_H-f_L, i.e. only the low-frequency second-order product is used and thus it is not useful to measure distortion produced by non-linear transfer functions which are symmetrical about zero. The following formula shows how CCIF2 IMD is calculated in Multi-Instrument.

$$\text{CCIF2 IMD} = \frac{V_{f_H-f_L}}{\mathrm{V}_{f_L} + \mathrm{V}_{f_H}} \times 100\%$$

where V_x is the RMS amplitude of the frequency component x.

IMD can also be expressed in dB:

$$(\text{CCIF2 IMD})_{\text{dB}} = 20\log_{10}(\text{CCIF2 IMD})$$

For example, if CCIF2 IMD = 0.0001%, then $(\text{CCIF2 IMD})_{\text{dB}}$ = -120 dB.

The CCIF2 IMD test signal can be generated using the MultiTone function of the Signal Generator in Multi-Instrument. Its configuration is shown in Fig. 8.

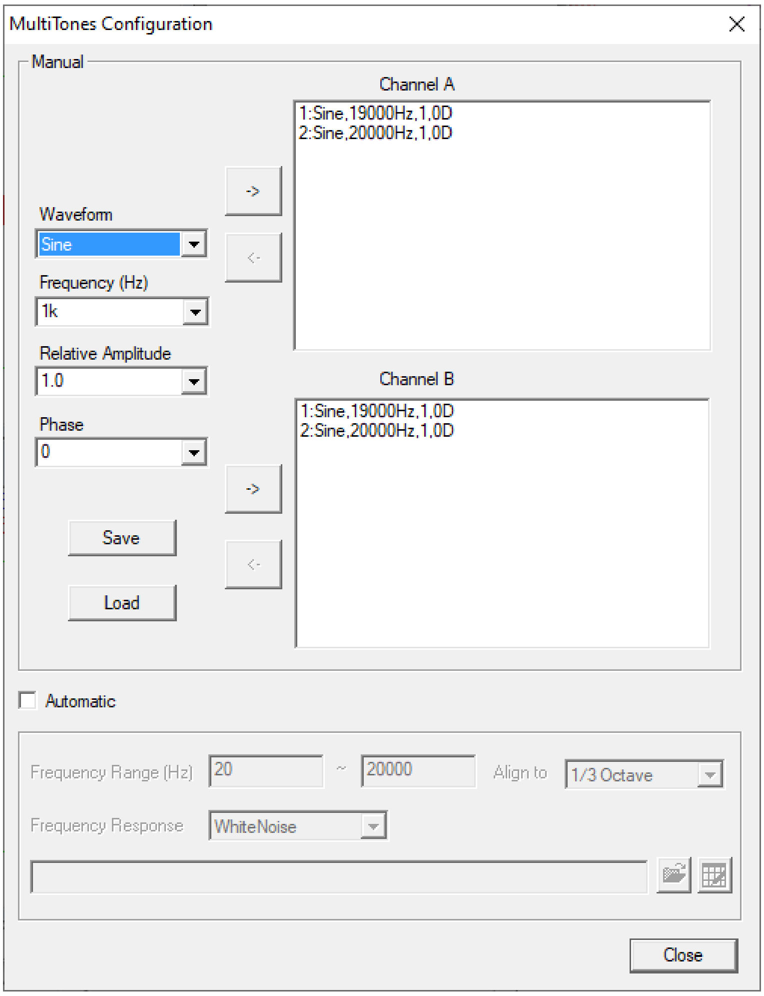

Fig. 8 CCIF2 IMD Test Signal Configuration

Fig. 9 shows the 24-bit software loopback test results of CCIF2 IMD. The ideal test signal is generated using the “iA=oA, iB=oB” software loopback mode of the Signal Generator of Multi-Instrument. The test parameters are: [Sampling Rate] = 48 kHz, [Frequency Composition] = 19000 Hz + 20000 Hz (with an amplitude ratio 1:1), [FFT Size] = 32768, [Sampling Bit Resolution] = 24, [Window Function] = Kaiser 8. No zero padding is applied as [Record Length] = 48000 which is greater than the FFT size. The intermodulation product used in CCIF2 IMD calculation is 1000Hz only. The measured CCIF2 IMD is 0.0000004% (-169.01 dB). As shown previously, the composite frequency of the standard CCIF2 IMD test signal is 1000 Hz. The greatest common factor of 1000 Hz and the sampling rate 48000 Hz is also 1000 Hz. Hence, the quantization noise is concentrated at 1000 Hz and its harmonic frequencies. The 1000 Hz frequency interval is not small and thus discrete frequency peaks at an interval of 1000 Hz can be clearly seen along the frequency axis. Some of the quantization noise is included in the CCIF2 IMD calculation. Nevertheless, this test shows that the residual CCIF2 IMD due to software is small enough for any practical CCIF2 IMD measurements. Furthermore, the noise in the hardware will help to randomize the 24-bit quantization noise, as shown in Fig. 10.

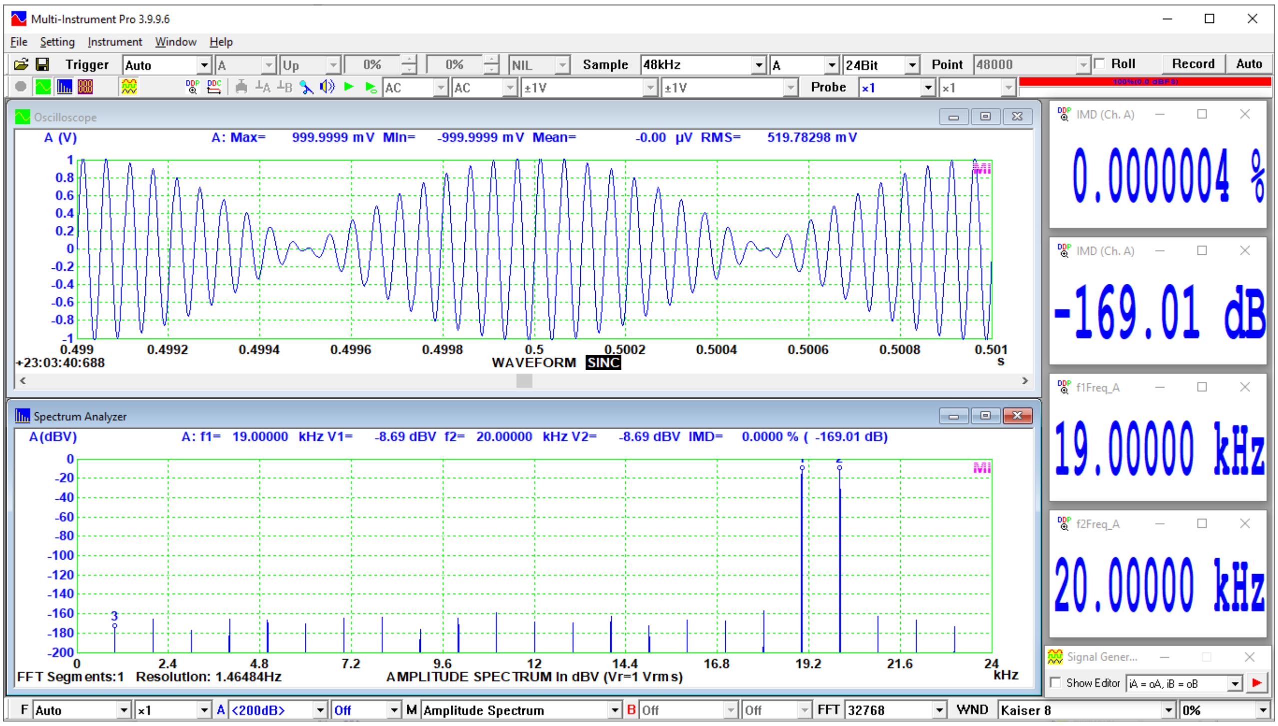

Fig. 9 CCIF2 IMD Software Loopback Test

Fig. 10 shows the 24-bit hardware loopback test results of CCIF2 IMD of a RTX6001 audio analyzer. The measured CCIF2 IMD is 0.0000568% (-124.91 dB).

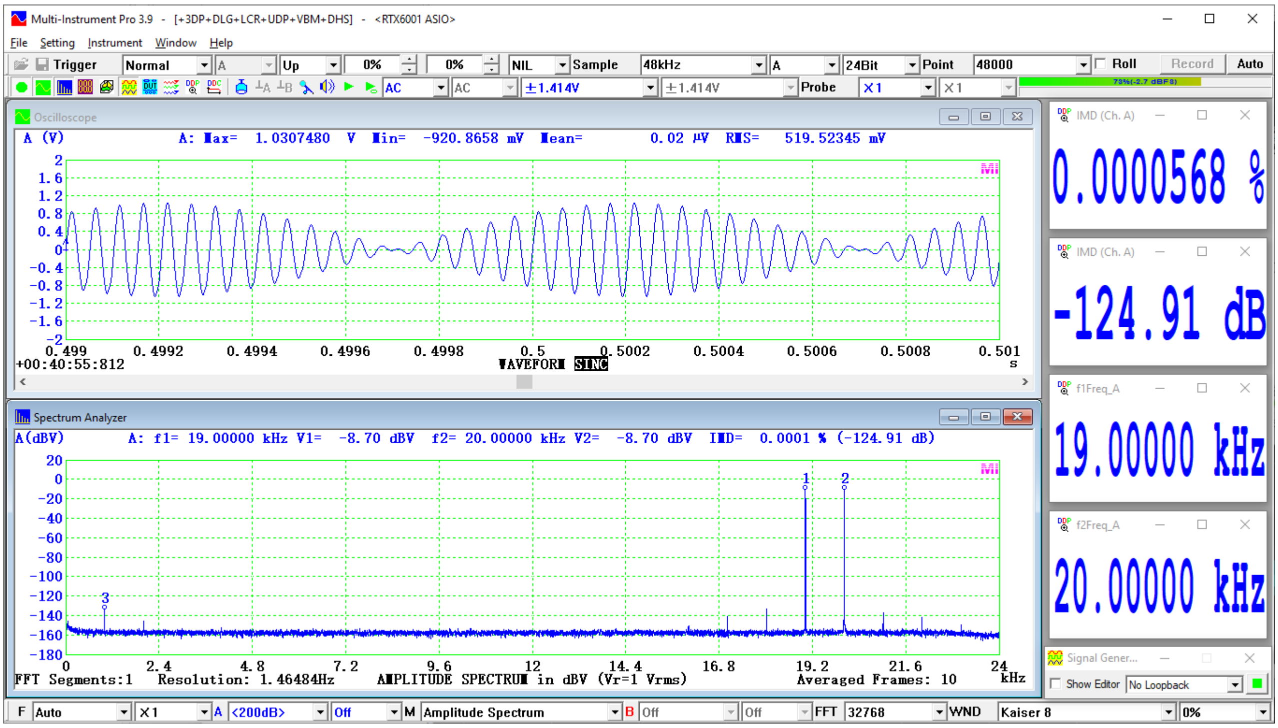

Fig. 10 CCIF2 IMD Hardware Loopback Test (RTX6001)

2.6.2 CCIF3 IMD Software and Hardware Loopback Tests

For CCIF3 IMD, the commonly used frequencies are: 13 kHz and 14 kHz, 14 kHz and 15 kHz, or 15 kHz and 16 kHz. The intermodulation distortion products used for this type of IMD calculation is: f_H-f_L, $2f_L$- f_H, $2f_H$-f_L, i.e. up to the third-order products are used. The following formula shows how CCIF3 IMD is calculated in Multi-Instrument.

$$\text{CCIF3 IMD} = \frac{\sqrt{V_{f_H-f_L}^2 + (V_{2f_L-f_H} + V_{2f_H-f_L})^2}}{V_{f_L} + V_{f_H}} \times 100\%$$

where V_x is the RMS amplitude of the frequency component x.

IMD can also be expressed in dB:

$$(\text{CCIF3 IMD})_{dB} = 20\log_{10}(\text{CCIF3 IMD})$$

For example, if CCIF3 IMD = 0.0001%, then $(\text{CCIF3 IMD})_{dB}$ = -120 dB.

The CCIF3 IMD test signal can be generated using the MultiTone function of the Signal Generator in Multi-Instrument. Its configuration is shown in Fig. 11.

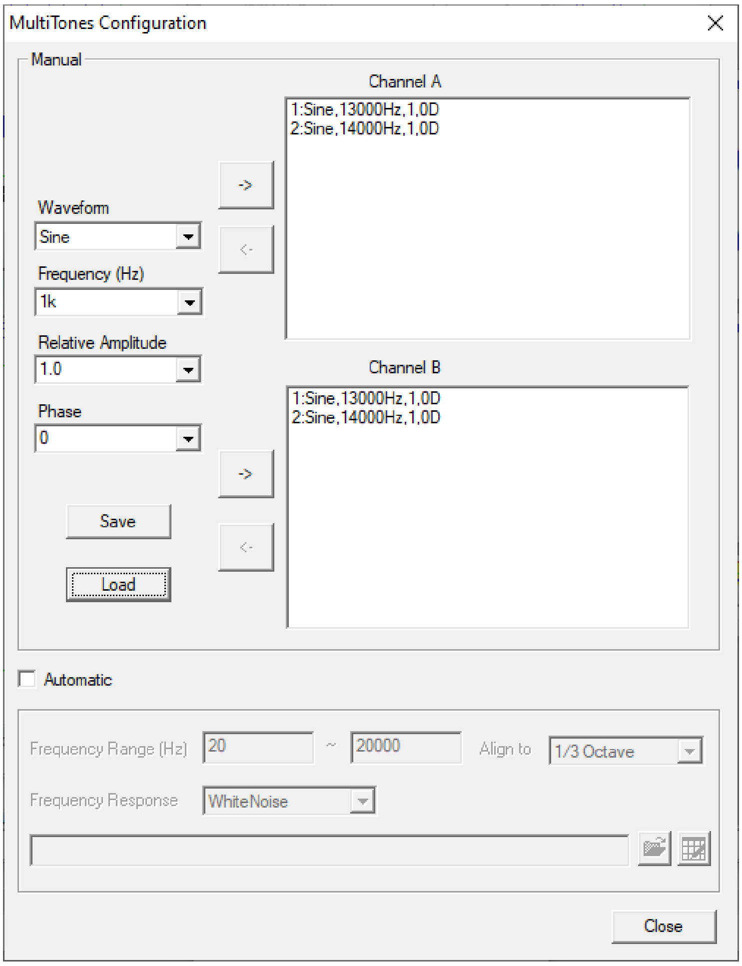

Fig. 11 CCIF3 IMD Test Signal Configuration

Fig. 12 shows the 24-bit software loopback test results of CCIF3 IMD. The ideal test signal is generated using the “iA=oA, iB=oB” software loopback mode of the Signal Generator of Multi-Instrument. The test parameters are: [Sampling Rate] = 48 kHz, [Frequency Composition] = 13000 Hz + 14000 Hz (with an amplitude ratio 1:1), [FFT Size] = 32768, [Sampling Bit Resolution] = 24, [Window Function] = Kaiser 8. No zero padding is applied as [Record Length] = 48000 which is greater than the FFT size. The intermodulation products used in this CCIF3 IMD calculation are 1000Hz, 12000Hz, and 15000Hz. The measured CCIF3 IMD is 0.0000028% (-151.17 dB). As shown previously, the composite frequency of the standard CCIF3 IMD test signal is 1000 Hz. The greatest common factor of 1000 Hz and the sampling rate 48000 Hz is also 1000 Hz. Hence, the quantization noise is concentrated at 1000 Hz and its harmonic frequencies. The 1000 Hz frequency interval is not small and thus discrete frequency peaks at an interval of 1000 Hz can be clearly seen along the frequency axis. Some of the quantization noise is included in the CCIF3 IMD calculation. Nevertheless, this test shows that the residual CCIF3 IMD due to software is small enough for any practical CCIF3 IMD measurements. Furthermore, the noise in the hardware will help to randomize the 24-bit quantization noise, as shown in Fig. 13.

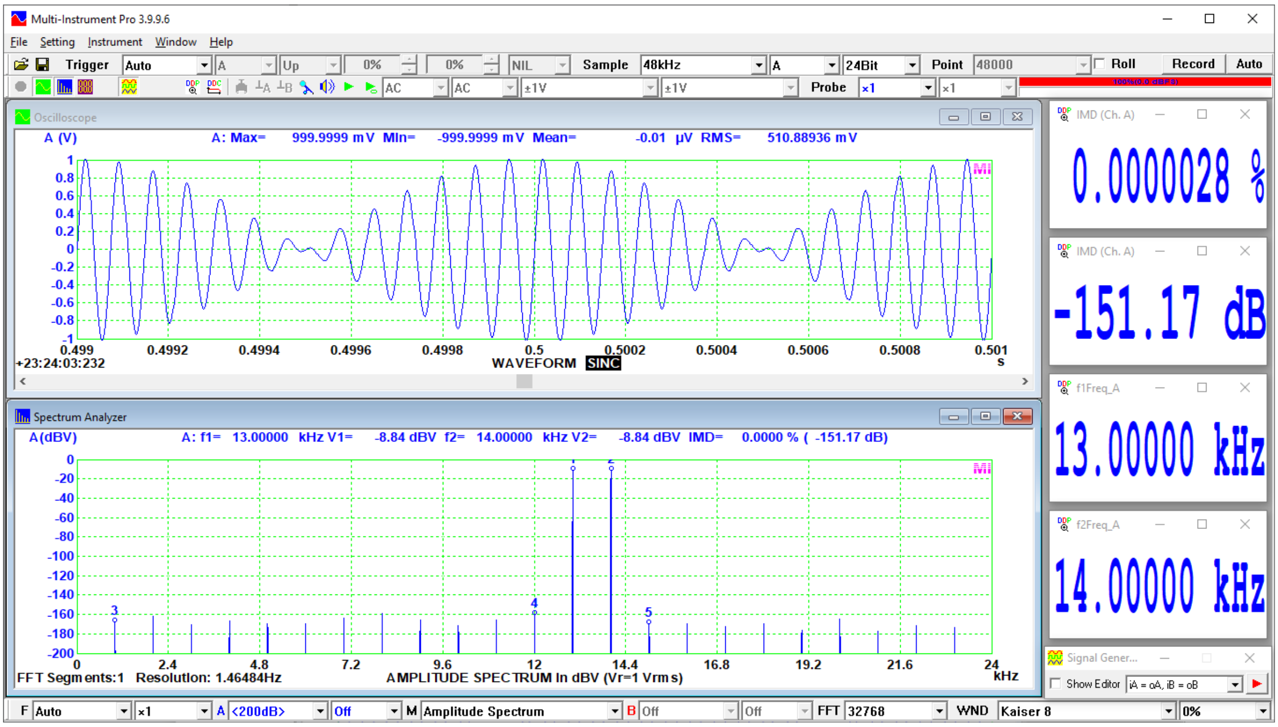

Fig. 12 CCIF3 IMD Software Loopback Test

Fig. 13 shows the 24-bit hardware loopback test results of CCIF3 IMD of a RTX6001 audio analyzer. The measured CCIF3 IMD is 0.000086% (-121.31 dB).

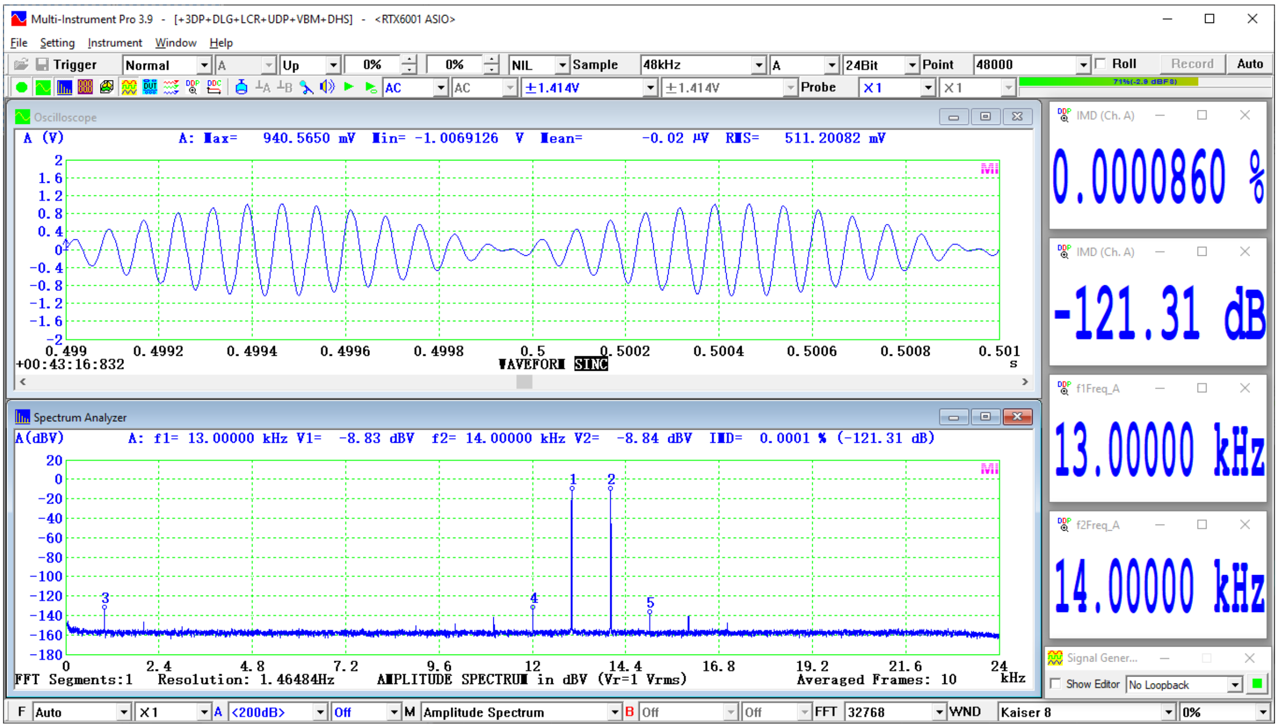

Fig. 13 CCIF3 IMD Hardware Loopback Test (RTX6001)

2.7 How to Randomize Quantization Noise

2.7.1 Slightly Adjusting Test Tone Frequencies

As mentioned previously, the IMD measurement error introduced by the correlation between the 24-bit quantization noise and the composite frequency of the two-tone signal can be ignored in most practical measurements, due to the existence of hardware noise which is usually sufficient to dither the 24-bit quantization process. However, this may not be true for the cases of 16-bit and 8-bit quantization.

Fig. 14 shows the 8-bit software loopback test results of DIN IMD. The two tones are 250 Hz and 8000 Hz respectively. The measured DIN IMD is 0.3588181% (-48.90 dB).

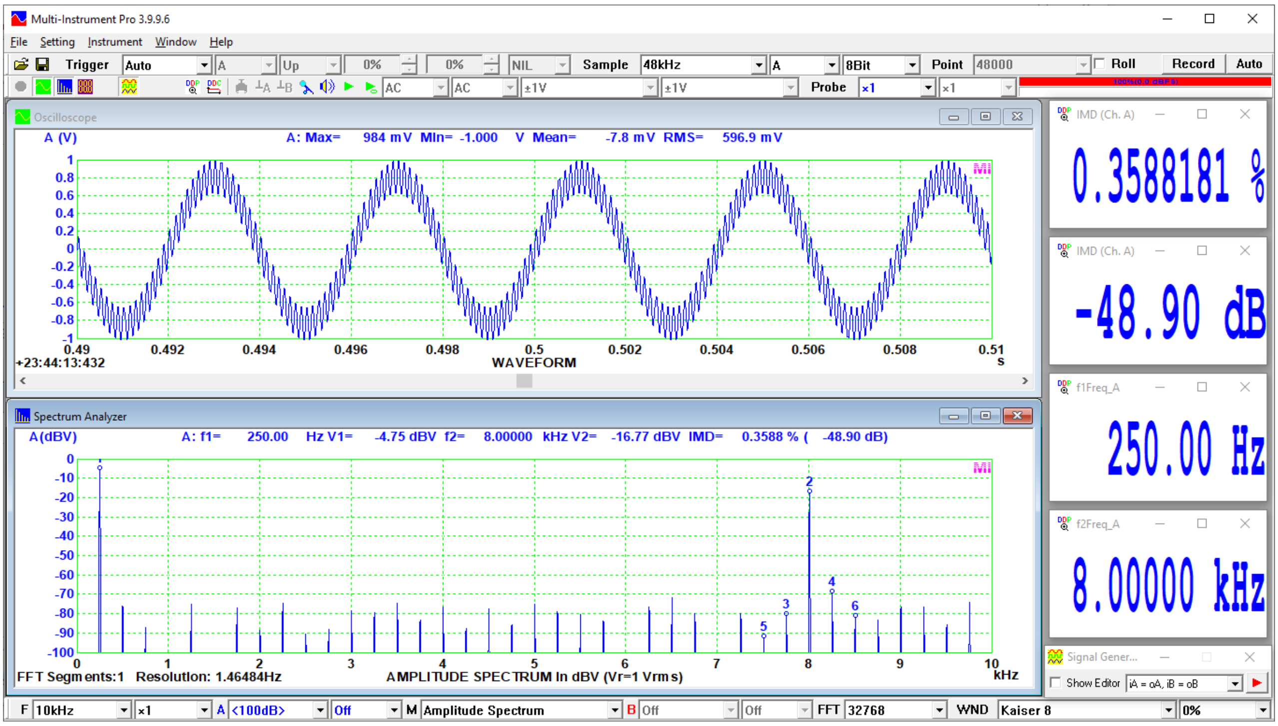

Fig. 14 DIN IMD Software Loopback Test (250Hz and 8000Hz, 8-bit)

If the high frequency 8000 Hz is changed to 8001 Hz, then the two tone frequencies: 250Hz and 8001Hz become coprime. The new composite frequency is 1 Hz. The greatest common factor f_{gcf} of 1 Hz and the sampling rate 48000 Hz is also 1 Hz. As the f_{gcf} is very small, the quantization noise will be sufficiently deconcentrated as shown in Fig. 15. The measured DIN IMD is 0.1500046% (-56.48 dB), which is about 8 dB lower than that in Fig. 14.

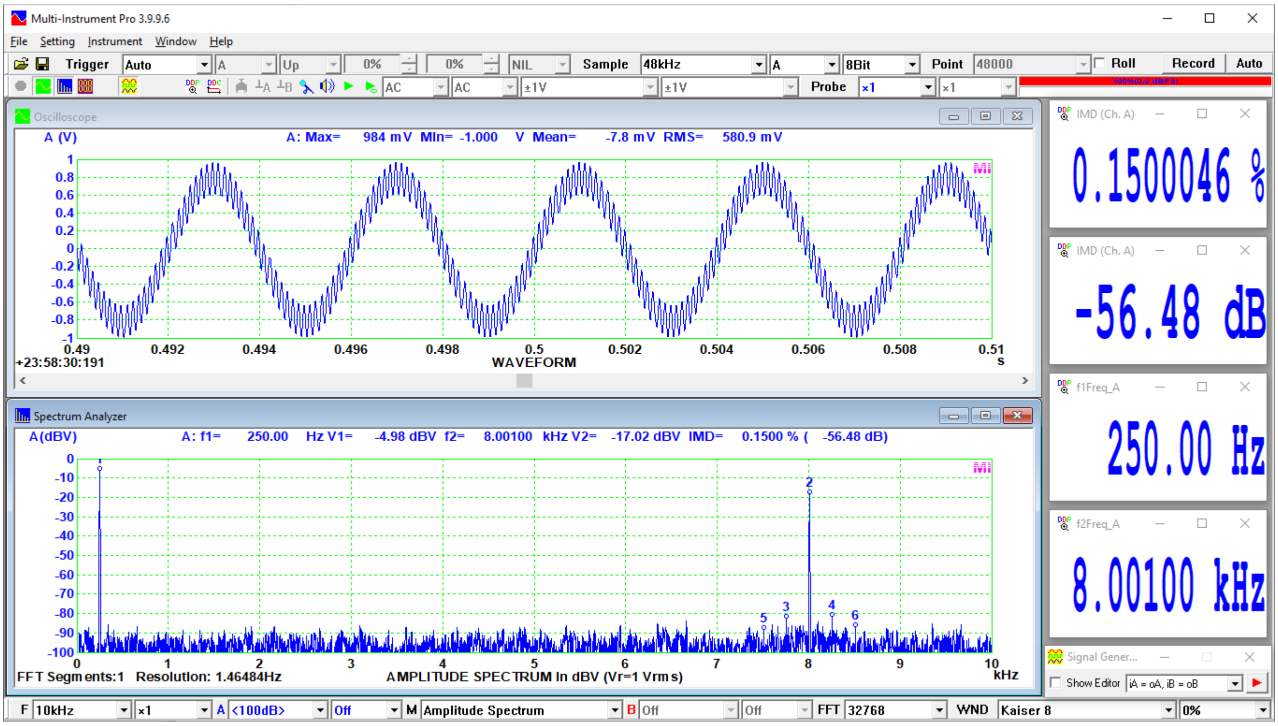

Fig. 15 DIN IMD Software Loopback Test (250Hz and 8001Hz, 8-bit)

2.7.2 Add Dither to Test Signal before Quantization

Adding a small amount of white noise (dither) with an amplitude of 0.5~1 bit to the two-tone test signal can also help to randomize the quantization noise. Fig.16 shows the configuration of a dithered DIN IMD test signal. The amplitude of the white noise is about 1/256 of that of the 250 Hz component, i.e. about 0.5 bit. Fig. 17 shows its 8-bit software loopback test results of DIN IMD. The measured DIN IMD is 0.1787465% (-54.96 dB), which is about 6 dB lower than that in Fig. 14.

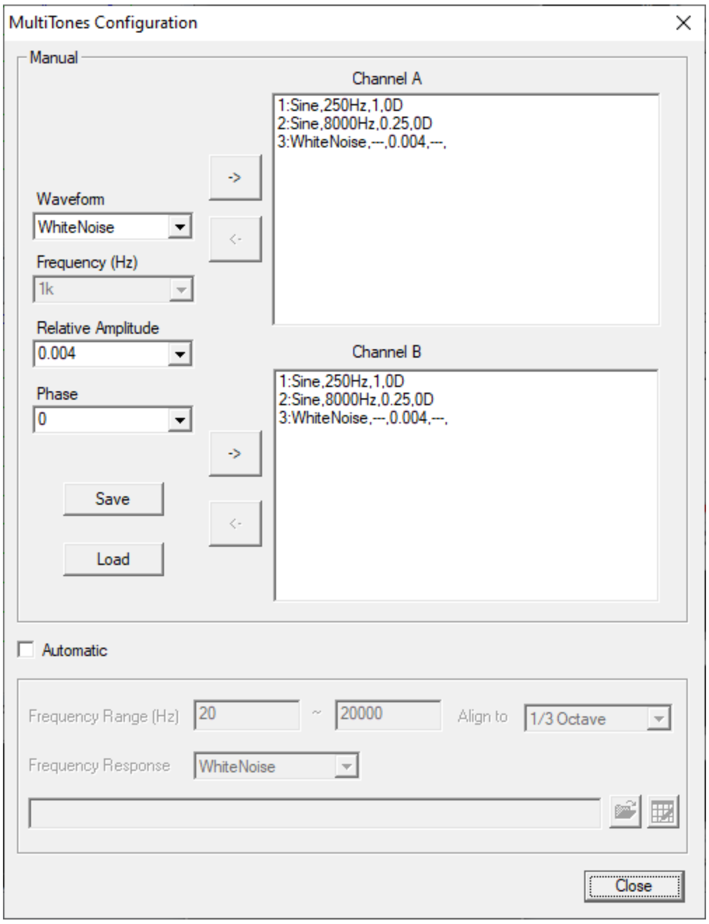

Fig. 16 Configuration of a dithered DIN IMD test signal

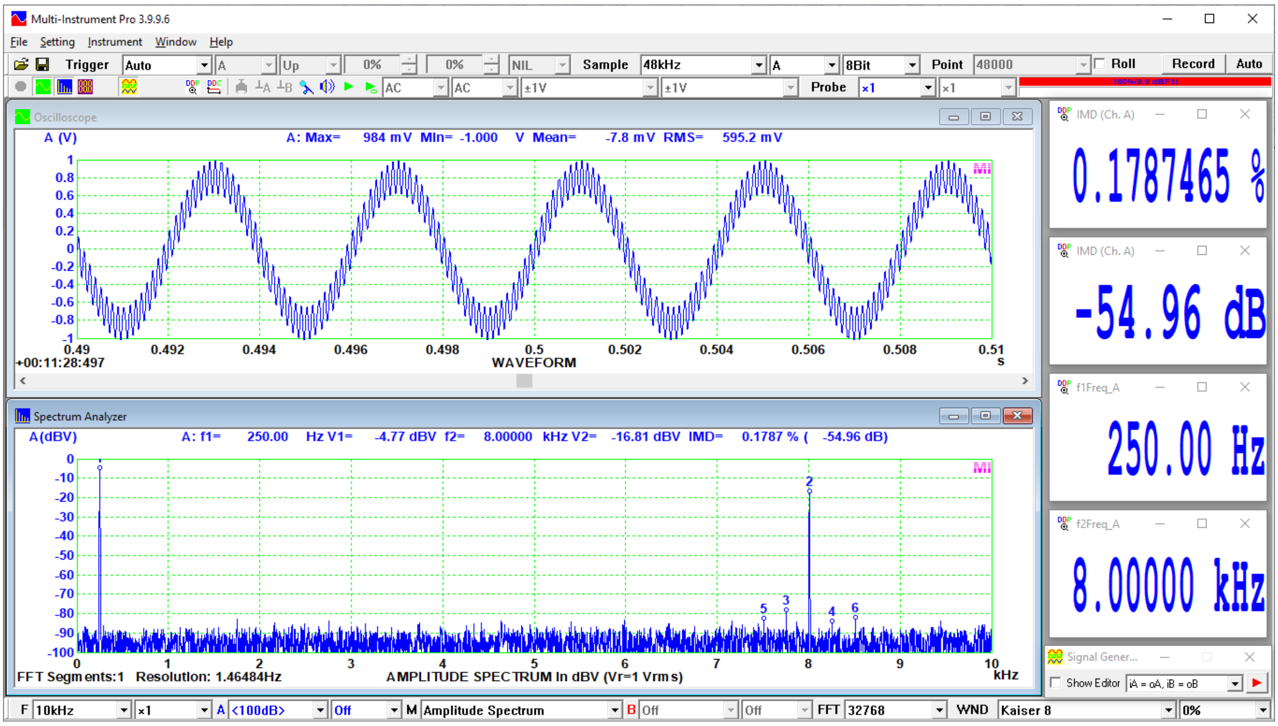

Fig. 17 DIN IMD Software Loopback Test (Dithered 250Hz and 8000Hz, 8-bit)

2.8 Some Sound Card Hardware Loopback Test Results

A professional sound card can be used as an IMD measuring device. Their performance can be verified through hardware loopback tests. The hardware loopback tests of some sound cards can be found in the following links.

(1) Focusrite Scarlett Solo

https://www.virtins.com/doc/Focusrite-Scarlett-Solo-Test-Report-using-Multi-Instrument.pdf

(2) EMU Tracker Pre

https://www.virtins.com/doc/D1004/EMU_Tracker_Pre_Report_D1004.pdf

(3) EMU 0204

https://www.virtins.com/doc/D1007/EMU_0204_Report.pdf

2.9 Estimation of Software Measurement Accuracy Using a Simulated Distortion Signal

As the performance of hardware continues to improve, it would be interest to see how the 24-bit quantization noise and numerical computation error start to affect the measurement accuracy. This can be evaluated using a simulated test signal. Fig. 18 shows the configuration of a multi-tone signal consisting of 60 Hz, 7000 Hz and 7060 Hz with an amplitude ratio of 1:0.25:0.0000001. Fig. 19 shows its software loopback test results. The measured SMPTE IMD is 0.0000411% (-127.72

dB) which is very close to its theoretical value 0.0000400% (-127.96 dB). The software measurement errors are thus negligibly small even at this low distortion level.

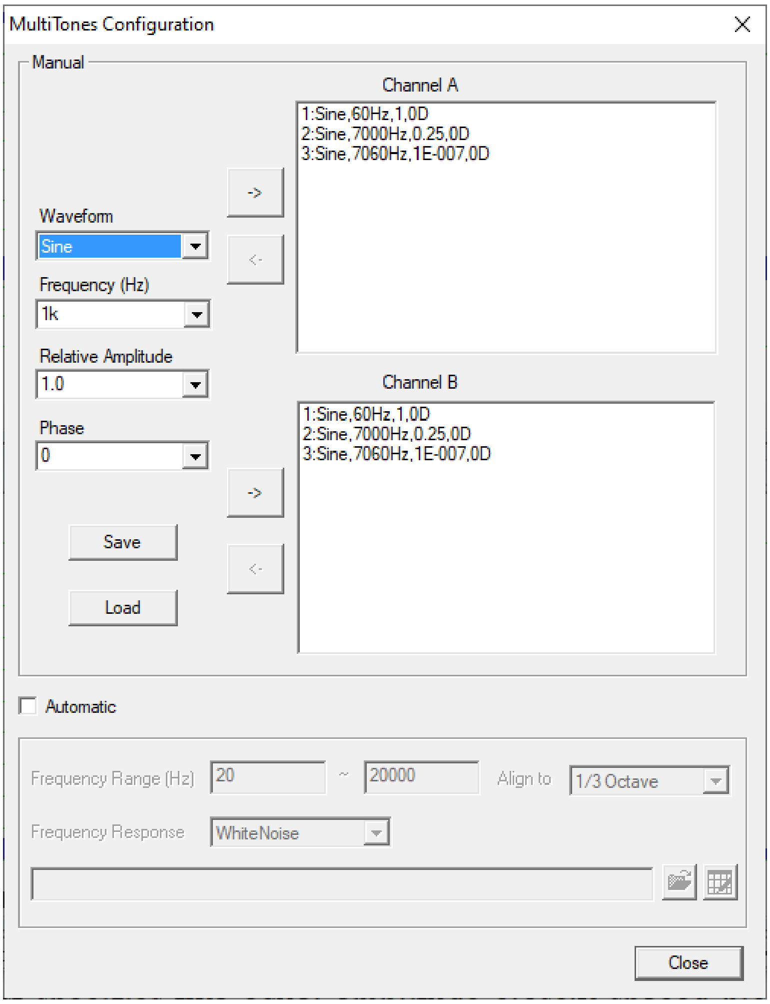

Fig. 18 Configuration of a SMPTE IMD test signal with a 0.00004% distortion

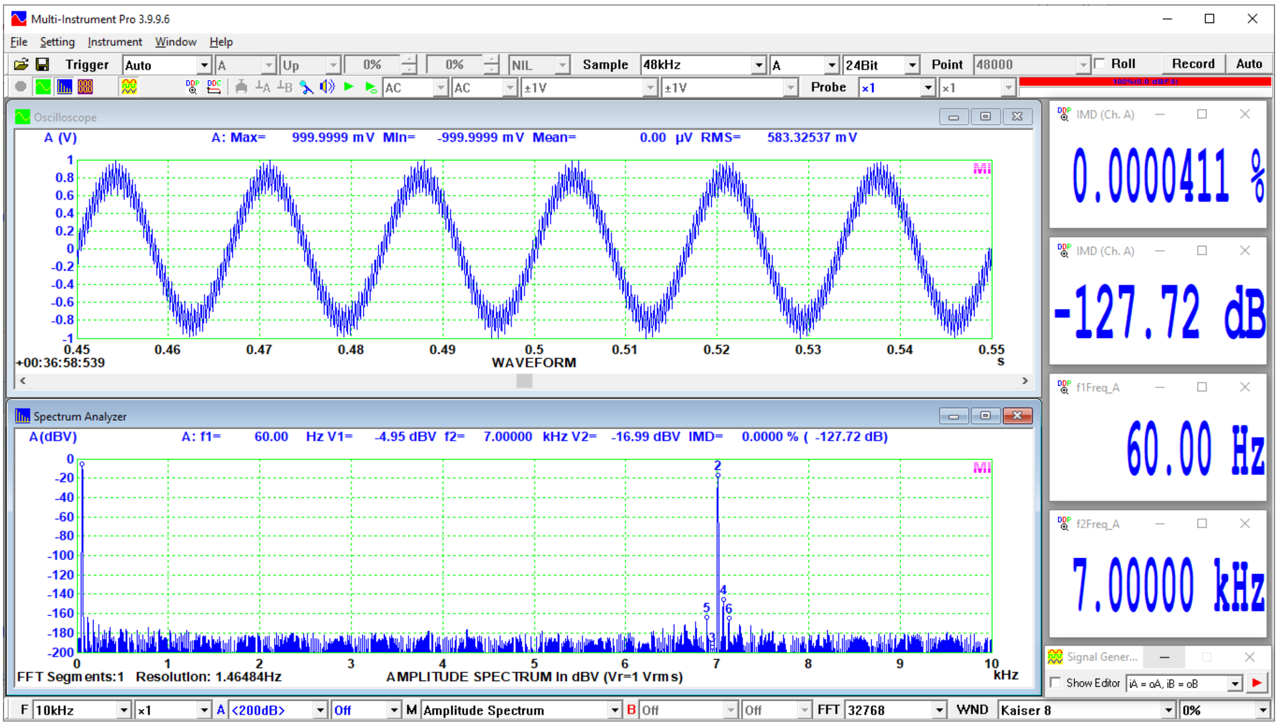

Fig. 19 Software Loopback Test of a SMPTE IMD test signal with a 0.00004% distortion

3 Multi-tone Total Distortion Plus Noise (TD+N)

Nonlinear distortion measurements depend heavily on the spectral content and level of the stimulus. Single-tone THD and Two-Tone IMD measurements are often criticized for being far too simple to represent a real-world audio signal such as music or speech. A multi-tone signal contains more than two frequencies and is a closer resemblance of real-world audio signals. It is usually used as a test signal for fast frequency response measurements (a.k.a linear distortion measurements). Meanwhile, it can also be used as a test signal for nonlinear distortion measurements. It excites both harmonic and intermodulation distortions in a DUT simultaneously. As the number of fundamentals, harmonics and their intermodulation products grows rapidly with the number of stimulus tones, it is extremely difficult to separate them. Thus the following method is used instead.

Total Distortion + Noise (TD+N) is defined as the square root of the ratio of the power of the total distortions plus noise to the power of all the fundamentals, in other words, the square root of the ratio of the total power less the power of all the fundamentals to the power of all the fundamentals. It can be expressed in percentage (%) or dB, as shown as follows:

$$\mathrm{TD+N} = \sqrt{\frac{\mathrm{V}_{Total}^2 - \sum_{\mathrm{i=1}}^{\mathrm{M}} \mathrm{V_i^2}}{\sum_{\mathrm{i=1}}^{\mathrm{M}} \mathrm{V_i^2}}} \times 100\%$$

$$(\mathrm{TD+N})_{\mathrm{dB}} = 20\log_{10}(\mathrm{TD+N})$$

where V_{total} is the RMS amplitude of the signal including all fundamentals, distortions and noise, V_i is the RMS amplitude of the i^{th} fundamental, M the total number of the fundamentals.

The frequency range of TD+N calculation can be set via [Spectrum Analyzer Processing]>"Parameter Measurement">"Range (Hz)". The default range is 20Hz~20kHz. Multi-Instrument detects the fundamentals in the multi-tone response automatically using its peak detection function. Therefore it is crucial that the "Number of Peaks" and "Deadband" are set correctly according to the stimulus's multi-tone configuration. When the stimulus contains only one frequency, then TD+N is equal to THD+N.

3.1 Multi-tone Configuration

The fundamental frequencies configured in the multi-tone stimulus signal should be carefully chosen such that they do not obviously coincide with their harmonics and intermodulation products. The number of stimulus tones should generally be kept below 32. These tones are typically spaced logarithmically across the audio frequency range and have equal amplitudes. Their phases should be selected to minimize the crest factor of the overall signal. This can usually be achieved by randomizing or collectively optimizing their initial phases.

3.2 Composite Frequency (or Repetition Rate) of a Multi-tone Signal

Assuming all the stimulus frequencies are integers and mutually irreducible, their composite frequency would be 1 Hz. Then the greatest common factor f_{gcf} of 1 Hz and any integer sampling rate would be 1 Hz as well. This is sufficient to deconcentrate the quantization noises. For a multi-tone signal with integer tone frequencies, it is almost impossible to use full-cycle sampling in a multi-tone measurement to avoid spectral leakage, thus window sampling has to be used to suppress it. Kaiser 6 ~ Kaiser 20, Blackman Harris 7, Cosine Sum 220, Cosine Sum 233, Cosine Sum 246, Cosine Sum 261 window functions are recommended. In Multi-Instrument, it is possible to tick a “No Spectral Leakage” option to allow the software to align each frequency configured in a multi-tone signal to the nearest FFT bin centerline frequency, which is usually a non-integer value. In such cases, Rectangle window should be employed.

3.3 Minimum Frequency Distance vs Real Frequency Resolution

The real frequency resolution of spectrum analysis must be finer than the minimum frequency distance between two adjacent stimulus frequencies in order to separate them in the spectrum. The real frequency resolution is equal to [Sampling Rate]/ [Record Length] if [Record Length]<[FFT Size] (i.e. with zero padding), and [Sampling Rate]/ [FFT Size] (i.e. without zero padding) if [Record Length]≥[FFT Size]. When there is no spectral leakage, a minimum frequency distance equal to the real frequency resolution is just enough to separate two adjacent fundamental frequencies completely. However, when a window function is used to suppress the spectral leakage, the minimum frequency distance required increases drastically and is window function dependent, usually in the range of 20~40 × [Real Frequency Resolution]. For example, about 20×[Real Frequency Resolution] should be used for Kaiser 8 window. Furthermore, it would be good to leave some space for noise and distortion within the minimum frequency distance. Hence, a minimum frequency distance of 50~100× [Real Frequency Resolution] is recommended for TD+N measurement. This requirement can be relaxed if a faster measurement time is preferred.

3.4 Software and Hardware Loopback Tests

The following multi-tone configuration is used here for the software and hardware loopback tests.

1:Sine,20Hz,1,0D
2:Sine,25Hz,1,0D
3:Sine,32Hz,1,0D
4:Sine,41Hz,1,0D
5:Sine,52Hz,1,0D
6:Sine,66Hz,1,0D
7:Sine,84Hz,1,0D
8:Sine,106Hz,1,0D
9:Sine,134Hz,1,0D
10:Sine,171Hz,1,0D
11:Sine,217Hz,1,0D
12:Sine,275Hz,1,0D
13:Sine,349Hz,1,0D
14:Sine,442Hz,1,0D
15:Sine,561Hz,1,0D
16:Sine,712Hz,1,0D
17:Sine,904Hz,1,0D
18:Sine,1147Hz,1,0D
19:Sine,1456Hz,1,0D
20:Sine,1847Hz,1,0D
21:Sine,2344Hz,1,0D
22:Sine,2975Hz,1,0D

23:Sine,3775Hz,1,0D
24:Sine,4790Hz,1,0D
25:Sine,6078Hz,1,0D
26:Sine,7713Hz,1,0D
27:Sine,9788Hz,1,0D
28:Sine,12420Hz,1,0D
29:Sine,15761Hz,1,0D
30:Sine,20000Hz,1,0D

The above multi-tone signal contains 30 frequencies logarithmically spaced across 20 Hz ~ 20 kHz. Each of them has a relative amplitude of 1 and an initial phase of 0 degree. The minimum frequency distance is 5 Hz, which is located at the low frequency end.

Fig. 20 shows the TD+N measurement settings for the above multi-tone stimulus. The frequency range is set to 15 Hz ~ 20005 Hz in order to take into account completely the energies of the 20 Hz and 20 kHz tones. The "Deadband" for peak detection is set to 4 Hz (just below the minimum frequency distance of 5 Hz) to avoid sub-peaks being falsely detected as peaks. The "number of peaks" is set to 30 to match the multi-tone configuration.

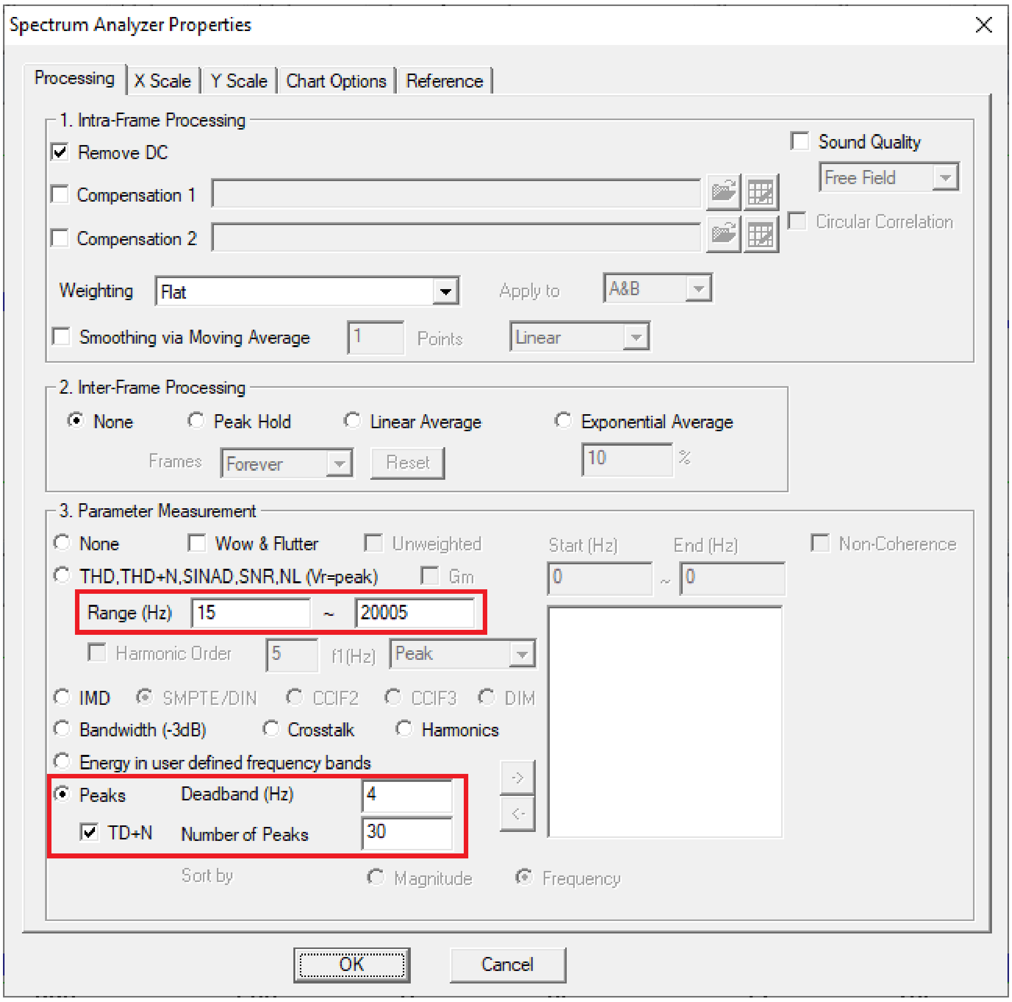

Fig. 20 TD+N Measurement Settings

Fig. 21 shows the 24-bit software loopback test results of TD+N of the above multi-tone stimulus. The ideal test data were generated by the Signal Generator of Multi-Instrument and saved into a

20-second WAV file. The file was then opened via [File]>[Open] for analysis. The measured TD+N is 0.0000188% (-134.53 dB). It is due to the numerical computation error and quantization noise. The real frequency resolution is 48000 / 960000 = 0.05 Hz and the apparent frequency resolution is 48000 / 1048576 =0.0457764 Hz. The minimum frequency distance of 5 Hz is then 5 / 0.05 =100 times as many as the real frequency resolution, leaving about 80% of the minimum frequency distance for noise and distortion. This is because Kaiser 8 window requires about 20 × [Real Frequency Resolution] to completely resolve the fundamental.

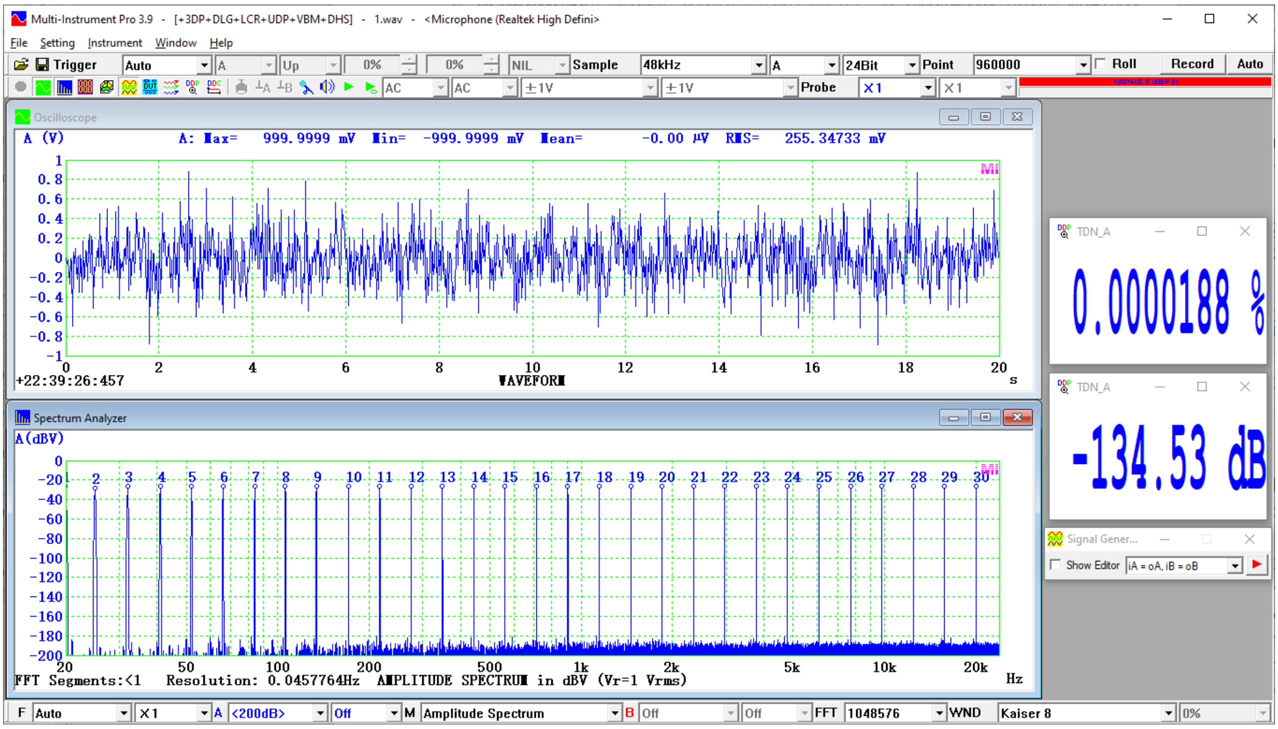

Fig. 21 TD+N Software Loopback Test

Fig. 22 shows the 24-bit hardware loopback test results of TD+N of a RTX6001 audio analyzer. The measured TD+N is 0.0006747% (-103.42 dB).

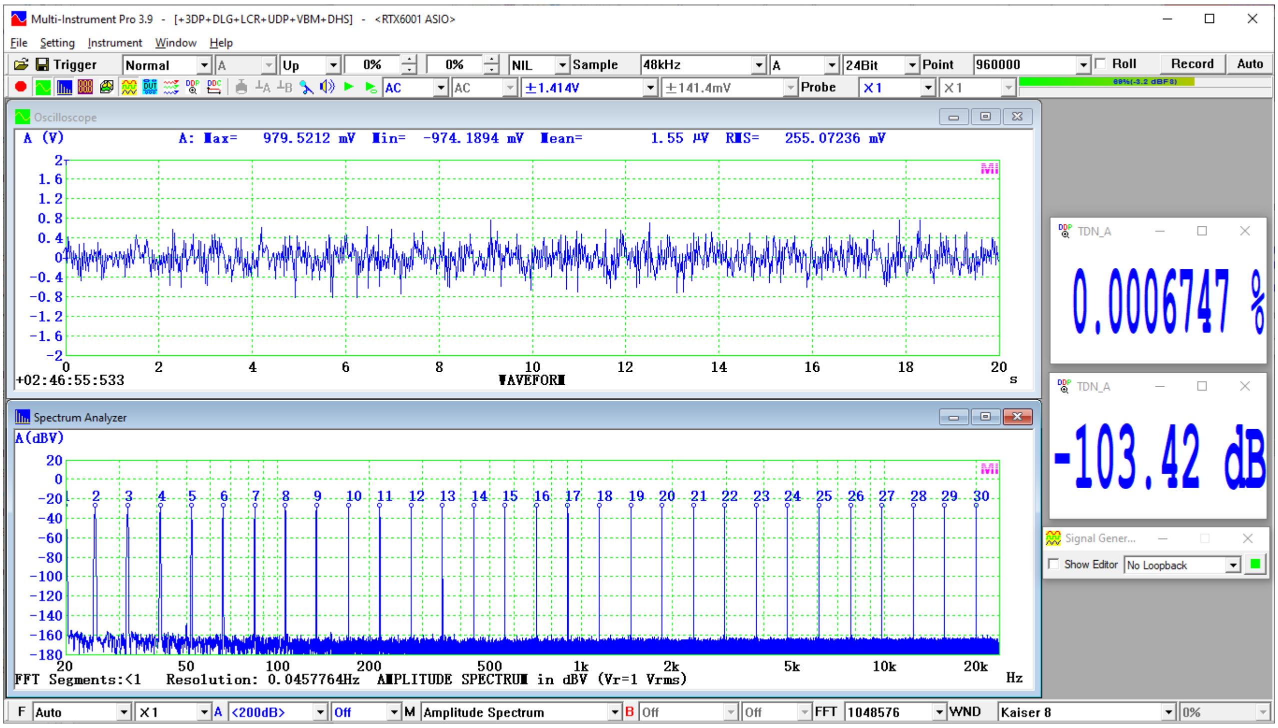

Fig. 22 TD+N Hardware Loopback Test (RTX6001)

3.5 Estimation of Software Measurement Accuracy Using a Simulated Distortion Signal

Fig. 23 shows the TD+N measurement results of a simulated distorted multi-tone signal. This signal contains the same 30 tones as those in the previous section. In addition, it has a 1000 Hz distortion component with a relative amplitude of 0.000005:

31:Sine,1000Hz,5E-006,0D

Its theoretical TD+N can be calculated as: $10\times\log_{10}(0.000005^2/(30\times1^2))$ = -120.79 dB. The measured value is -120.49 dB (0.0000945%) which is very close to the theoretical value. The software measurement errors are thus negligibly small even at this low distortion level.

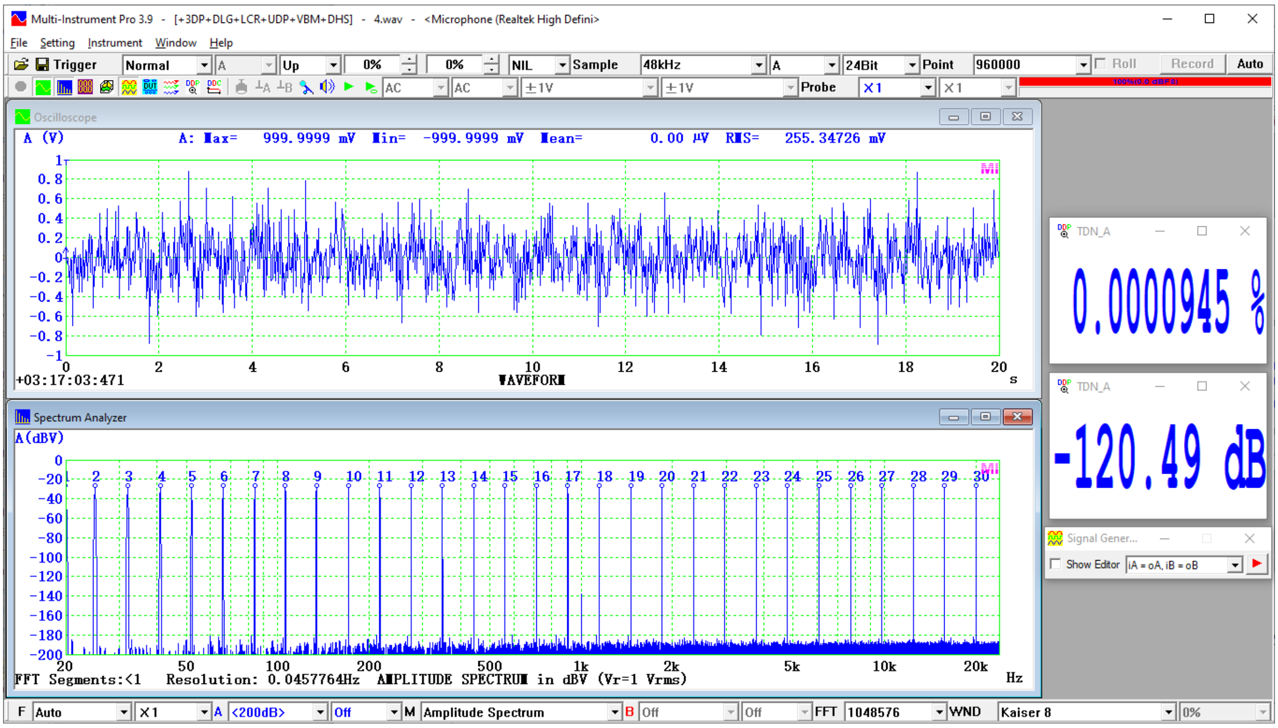

Fig. 23 Software Loopback Test of a TD+N test signal with a -120.79 dB distortion

4 Dynamic Intermodulation Distortion (DIM) / Transient Intermodulation Distortion (TIM)

Nonlinear distortions, particularly in power amplifiers, can be categorized as either static or dynamic. Those that result from basic device nonlinearity are called static distortions. They are not really frequency dependent although their magnitude may change with the change in feedback factor with frequency. Thus, the distortions vary with frequency in direct proportion to the change in feedback with frequency. This distinguishes such distortions from ones which inherently change magnitude with frequency - often called "dynamic" distortions. Since dynamic distortions change with frequency, even without a change in feedback with frequency, the addition of frequency dependent feedback creates a high order dependence of distortion on frequency. The term dynamic distortion has commonly referred to mechanisms which get worse with fast or high frequency signals. In other words, dynamic distortion is very much slew rate related, and thus it is sometimes called Slewing Induced Distortion (SID) or Slope Induced Distortion (SID), Transient Intermodulation Distortion (TIM). It is excited by the signal rate of change and becomes worse when slew rate limiting occurs.

Numerous techniques have been proposed for measuring static and dynamic distortions. Measuring THD as a function of frequency and level yields a complete performance profile of a DUT. For those severely bandlimited DUTs, other techniques must be employed. If static distortions are of special interest, SMPTE/DIN IMD can be used. If dynamic distortions are of concern, CCIF should be measured. If the DUT's bandwidth allows, measuring DIM is an effective and efficient way to characterize both the static and dynamic distortions. It is designed to be particularly sensitive to distortions produced during transient conditions. According to IEC 60268-3, the DIM test signal consists of a bandlimited square wave component at a frequency of 3.15 kHz and a sine wave component at a frequency of 15 kHz with an amplitude of ¼ of that of the square wave component. Two DIM tests are common: DIM30 and DIM100. DIM30 uses a single-pole low-pass filter with a cutoff frequency of 30 kHz while in DIM100 the cutoff frequency is 100 kHz. In both tests, nine intermodulation products are measured. They are listed in the following table.

Intermodulation (f_q=3.15 kHz, f_s=15 kHz)	Frequency (kHz)	Symbol
$5f_q - f_s$	0.75	U1
$f_s - 4f_q$	2.40	U2
$6f_q - f_s$	3.90	U3
$f_s - 3f_q$	5.55	U4
$7f_q - f_s$	7.05	U5
$f_s - 2f_q$	8.70	U6
$8f_q - f_s$	10.20	U7
$f_s - f_q$	11.85	U8
$9f_q - f_s$	13.35	U9

DIM is calculated as the square root of the ratio of the power of the above nine intermodulation products to the power of the 15 kHz sine wave. It can be expressed in percentage (%) or dB, as shown as follows:

$$\text{DIM} = \frac{\sqrt{\sum_{i=1}^{9} V_i^2}}{V_s} \times 100\%$$

$$(\text{DIM})_{\text{dB}} = 20\log_{10}(\text{DIM})$$

where V_i is the RMS amplitude of the i[th] intermodulation product and V_s the RMS amplitude of the 15kHz sine wave.

4.1 DIM Test Signal Generation

DIM test signal can be generated using the Multitone function of the Signal Generator in Multi-Instrument. The simplest configuration would be adding a 3150 Hz square wave with a relative amplitude of 1 and a 15000 Hz sine wave with a relative amplitude of 0.25 together as follows:

1:Rectangle,3150Hz,1,0D
2:Sine,15000Hz,0.25,0D

However, the above configuration requires an external 30 kHz (for DIM30) or 100 kHz (for DIM100) single-pole low pass filter to band-limit the signal. Also, the square wave contains an infinite number of harmonics which may cause aliasing during DAC and ADC if the harmonics above ½ sampling rate are not filtered out completely. These two issues can be solved by pre-filtering the test signal.

An ideal square wave *x(t)* with an amplitude of 1 can be decomposed into a sum of a series of sine waves as follows:

$$x(t) = \frac{4}{\pi} \sum_{n=1,3,5,\ldots}^{\infty} \frac{1}{n} \sin(2n\pi f t)$$

The above equation shows that the amplitude of the square wave is π/4 of that of its fundamental. If the amplitude of the fundamental is 1, then the amplitude of the square wave is π/4. The amplitude of the 15 kHz sine wave in the DIM test signal is thus π /4/4 = 0.196350. With the above information, we can configure sharply bandlimited DIM test signals for DIM30 and DIM100 respectively as follows.

(1) DIM30_SharplyBandLimited
1:Sine,3150Hz,1,0D
2:Sine,9450Hz,0.333333,0D
3:Sine,15750Hz,0.2,0D
4:Sine,22050Hz,0.142857,0D
5:Sine,28350Hz,0.111111,0D
6:Sine,15000Hz,0.196350,0D

The square wave component is sharply bandlimited by a 30 kHz brick-wall (ideal) low-pass filter. An output sampling rate of at least 96 kHz and an analog bandwidth of at least 30 kHz are recommended for signal generation.

(2) DIM100_SharplyBandLimited
1:Sine,3150Hz,1,0D
2:Sine,9450Hz,0.333333,0D
3:Sine,15750Hz,0.2,0D
4:Sine,22050Hz,0.142857,0D
5:Sine,28350Hz,0.111111,0D
6:Sine,34650Hz,0.090909,0D
7:Sine,40950Hz,0.076923,0D
8:Sine,47250Hz,0.066667,0D
9:Sine,53550Hz,0.058824,0D
10:Sine,59850Hz,0.052632,0D
11:Sine,66150Hz,0.047619,0D
12:Sine,72450Hz,0.043478,0D
13:Sine,78750Hz,0.04,0D
14:Sine,85050Hz,0.037037,0D
15:Sine,91350Hz,0.034483,0D
16:Sine,15000Hz,0.196350,0D

The square wave component is sharply bandlimited by a 100 kHz brick-wall (ideal) low-pass filter. An output sampling rate of at least 192 kHz and an analog bandwidth of at least 96 kHz are recommended for signal generation.

Using a sharply bandlimited DIM test signal saves some analog bandwidth for the measuring system. If the analog bandwidth is enough, then we can use single-pole low-pass filtered DIM test signals. The gain of a single-pole low-pass filter is given by:

$$Gain(f) = \frac{1}{\sqrt{1 + \left(\frac{f}{f_c}\right)^2}}$$

where f is the frequency, f_c the cut-off frequency of the low-pass filter. The single-pole low-pass filtered DIM test signals can then be configured as follows.

(3) DIM30_SinglePole_LowpassFiltered
1:Sine,3150Hz,0.994533,0D
2:Sine,9450Hz,0.317933,0D
3:Sine,15750Hz,0.177080,0D
4:Sine,22050Hz,0.115109,0D
5:Sine,28350Hz,0.080757,0D
6:Sine,34650Hz,0.059505,0D
7:Sine,40950Hz,0.045460,0D
8:Sine,47250Hz,0.035734,0D
9:Sine,53550Hz,0.028750,0D
10:Sine,59850Hz,0.023585,0D
11:Sine,66150Hz,0.019668,0D
12:Sine,72450Hz,0.016634,0D
13:Sine,78750Hz,0.014240,0D
14:Sine,85050Hz,0.012320,0D
15:Sine,91350Hz,0.010759,0D

16:Sine,15000Hz,0.196350,0D

The square wave component is bandlimited by a 30 kHz single-pole (first-order) low-pass filter. An output sampling rate of at least 192 kHz and an analog bandwidth of at least 96 kHz are recommended for signal generation.

(4) DIM100_SinglePole_LowpassFiltered
1:Sine,3150Hz,0.999504,0D
2:Sine,9450Hz,0.331855,0D
3:Sine,15750Hz,0.197565,0D
4:Sine,22050Hz,0.139506,0D
5:Sine,28350Hz,0.106898,0D
6:Sine,34650Hz,0.085899,0D
7:Sine,40950Hz,0.071186,0D
8:Sine,47250Hz,0.060277,0D
9:Sine,53550Hz,0.051856,0D
10:Sine,59850Hz,0.045161,0D
11:Sine,66150Hz,0.039716,0D
12:Sine,72450Hz,0.035209,0D
13:Sine,78750Hz,0.031425,0D
14:Sine,85050Hz,0.028213,0D
15:Sine,91350Hz,0.025459,0D
16:Sine,97650Hz,0.023079,0D
17:Sine,103950Hz,0.021009,0D
18:Sine,110250Hz,0.019195,0D
19:Sine,116550Hz,0.017599,0D
20:Sine,122850Hz,0.016187,0D
21:Sine,129150Hz,0.014932,0D
22:Sine,135450Hz,0.013813,0D
23:Sine,141750Hz,0.012810,0D
24:Sine,148050Hz,0.011909,0D
25:Sine,154350Hz,0.011097,0D
26:Sine,160650Hz,0.010362,0D
27:Sine,166950Hz,0.009695,0D
28:Sine,173250Hz,0.009089,0D
29:Sine,179550Hz,0.008536,0D
30:Sine,185850Hz,0.008031,0D
31:Sine,15000Hz,0.196350,0D

The square wave component is bandlimited by a 100 kHz single-pole (first-order) low-pass filter. An output sampling rate of at least 384 kHz and an analog bandwidth of at least 192 kHz are recommended for signal generation.

4.2 Composite Frequency (or Repetition Rate) of DIM Test Signal

The composite frequency of the 3150 Hz square wave (or its bandlimited counterpart) and the 15000 Hz sine wave is 150 Hz. Given the commonly used sampling rates: 44.1 kHz, 48 kHz, 50 kHz, 96 kHz, 100 kHz, 192 kHz, 200 kHz, etc., it is almost impossible to use full-cycle sampling with FFT. Therefore, spectral leakage is inevitable and a window function must be used to suppress it. For DIM measurement, Kaiser 6 ~ Kaiser 20, Blackman Harris 7, Cosine Sum 220, Cosine Sum 233, Cosine Sum 246, Cosine Sum 261 window functions are recommended.

4.3 Software and Hardware Loopback Tests

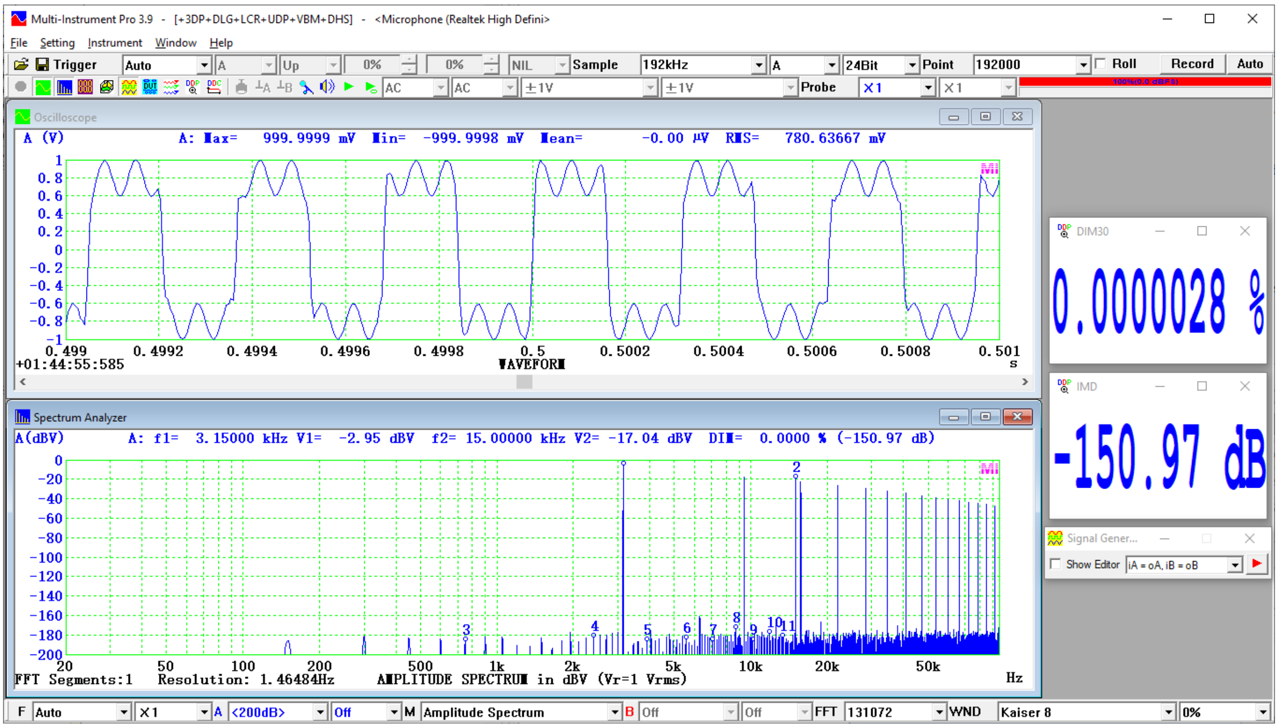

Fig. 24 DIM 30 Software Loopback Test (Single-pole low-pass filtered 3.15 kHz square wave and 15 kHz sine wave, amplitude ratio 4:1)

The 24-bit software loopback test results of DIM30 using the aforementioned single-pole low-pass filtered DIM test signal are shown in Fig. 24. As the greatest common factor f_{gcf} of the composite frequency 150 Hz and the sampling rate 48000 Hz is 150 Hz, the 150 Hz fundamental and its harmonics resulting from the quantization noise can be clearly observed in the figure. The residual DIM is only 0.0000028% (-150.97 dB), which is negligibly small and the hardware noise is usually sufficient to randomize the 24-bit quantization noise in the actual measurements. On the other hand, it is possible randomize the quantization noise by, for example, changing the 15000 Hz sine wave to 150001Hz (see Fig. 25) or adding a 0.5~1 bit white noise component in the test signal.

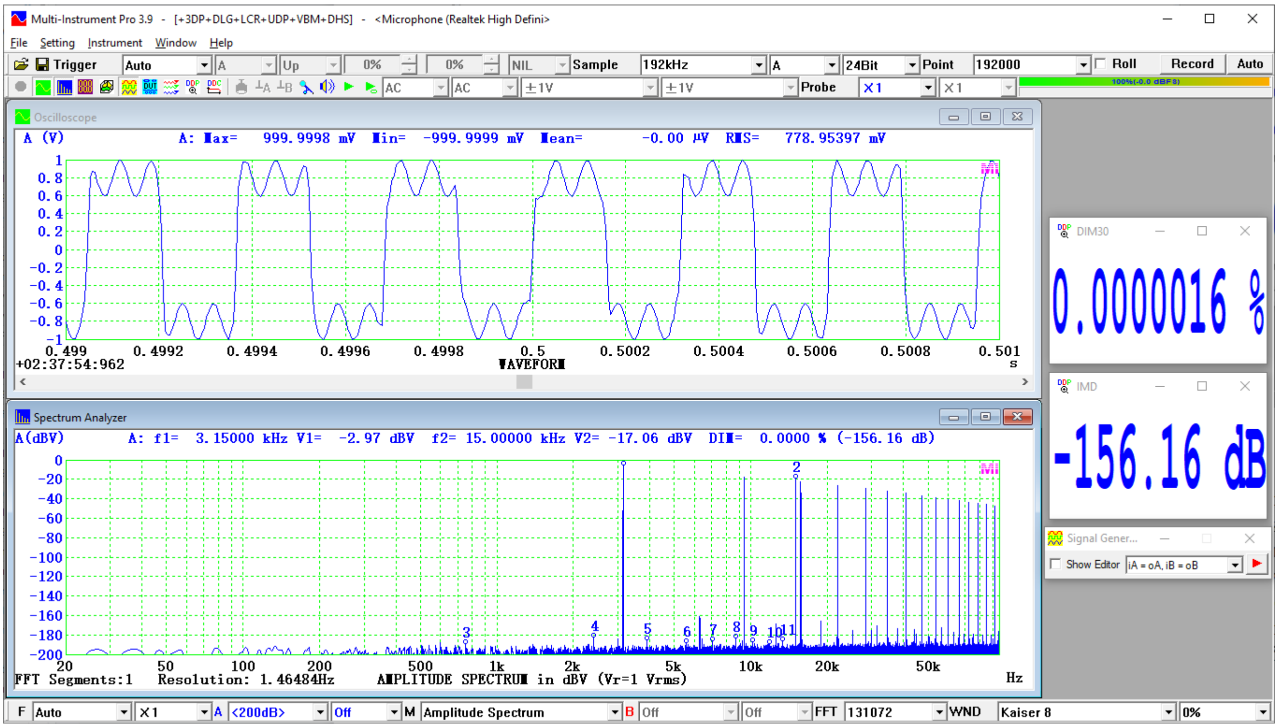

Fig. 25 DIM 30 Software Loopback Test (Single-pole low-pass filtered 3.15 kHz square wave and 15.001 kHz sine wave, amplitude ratio 4:1)

Fig. 26 shows the 24-bit hardware loopback test results of DIM30 of a RTX6001 audio analyzer. The measured DIM 30 is 0.0001495% (-116.51 dB).

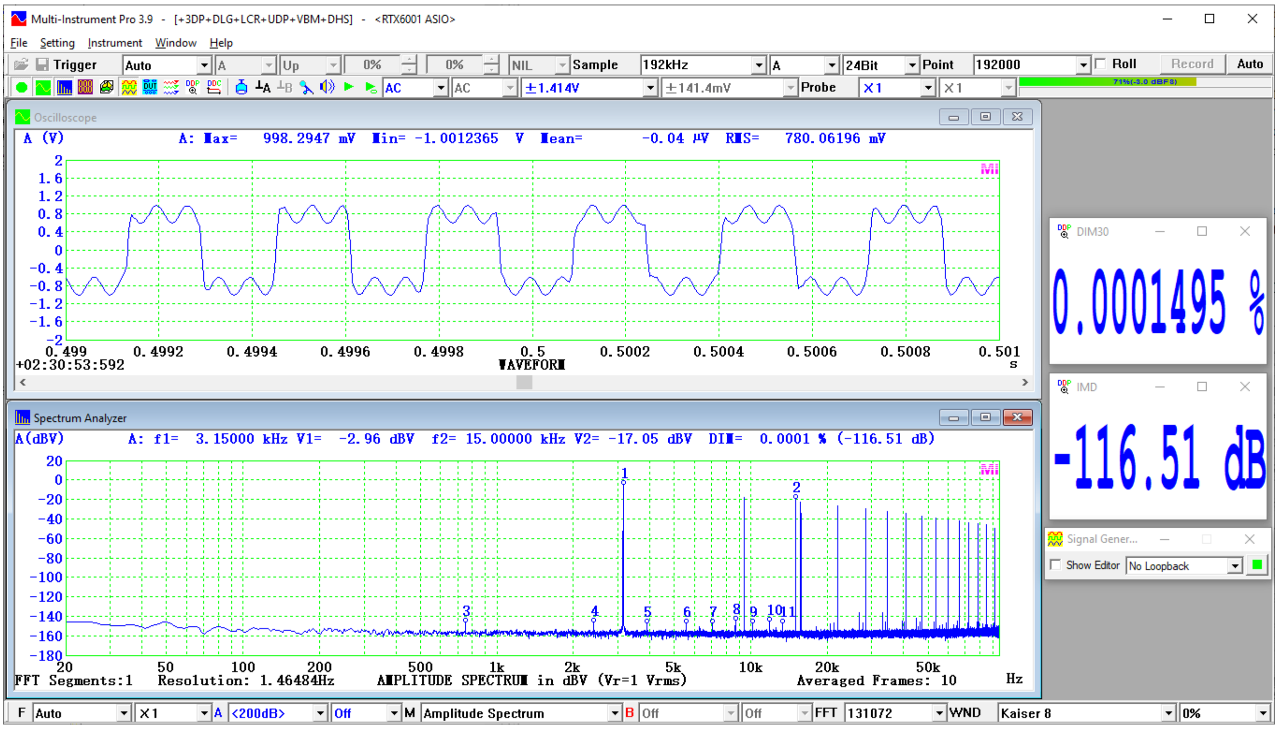

Fig. 26 DIM 30 Hardware Loopback Test of RTX6001 (Single-pole low-pass filtered 3.15 kHz square wave and 15 kHz sine wave, amplitude ratio 4:1)

4.4 Estimation of Software Measurement Accuracy Using a Simulated Distortion Signal

Fig. 27 shows the DIM30 measurement results of a simulated distorted DIM30 test signal. This signal contains the same 16 tones as those in the aforementioned DIM30_SinglePole_LowpassFiltered test signal. In addition, it has a 750 Hz distortion component with a relative amplitude of 0.000000019635:

17:Sine,750Hz,1.9635E-008,0D

Its theoretical DIM30 can be calculated as: 1.9635E-008/0.19635 = -0.00001% (-140.00 dB). The measured value is -0.0000108% (-139.31 dB) which is very close to the theoretical value. The software measurement errors are thus negligibly small even at this low distortion level.

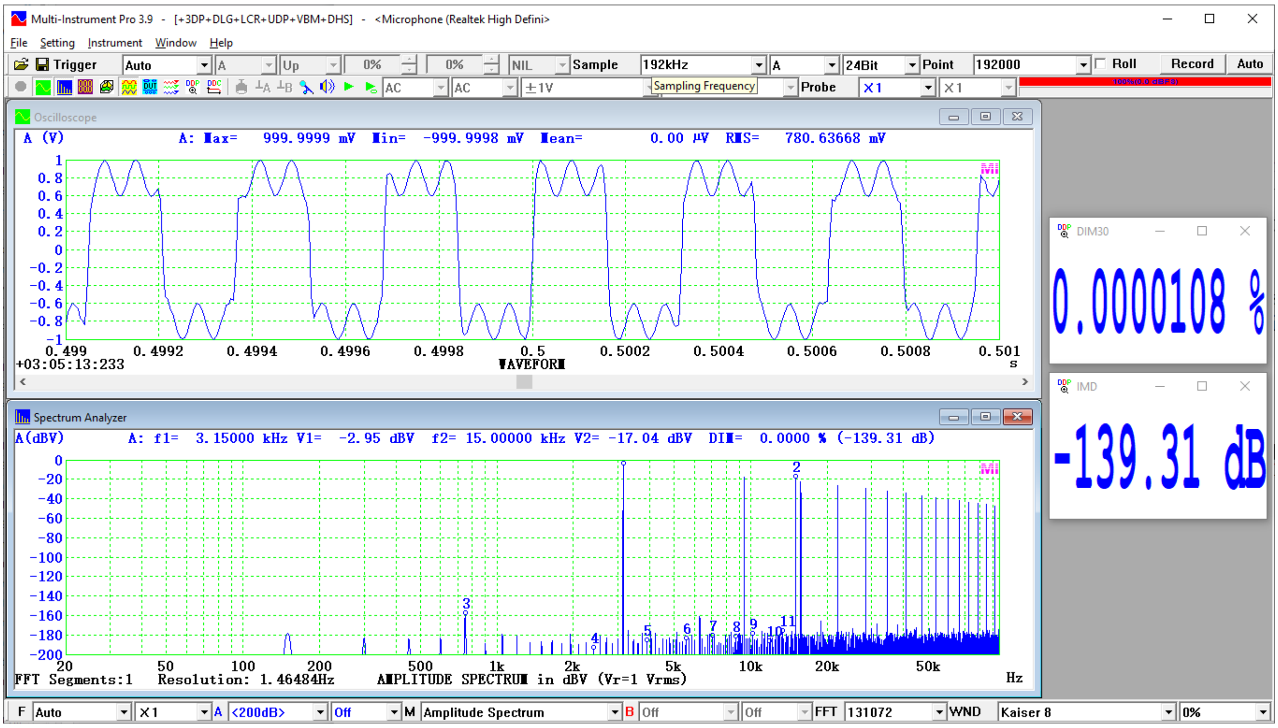

Fig. 27 Software Loopback Test of a DIM30 test signal with a -140.00 dB distortion