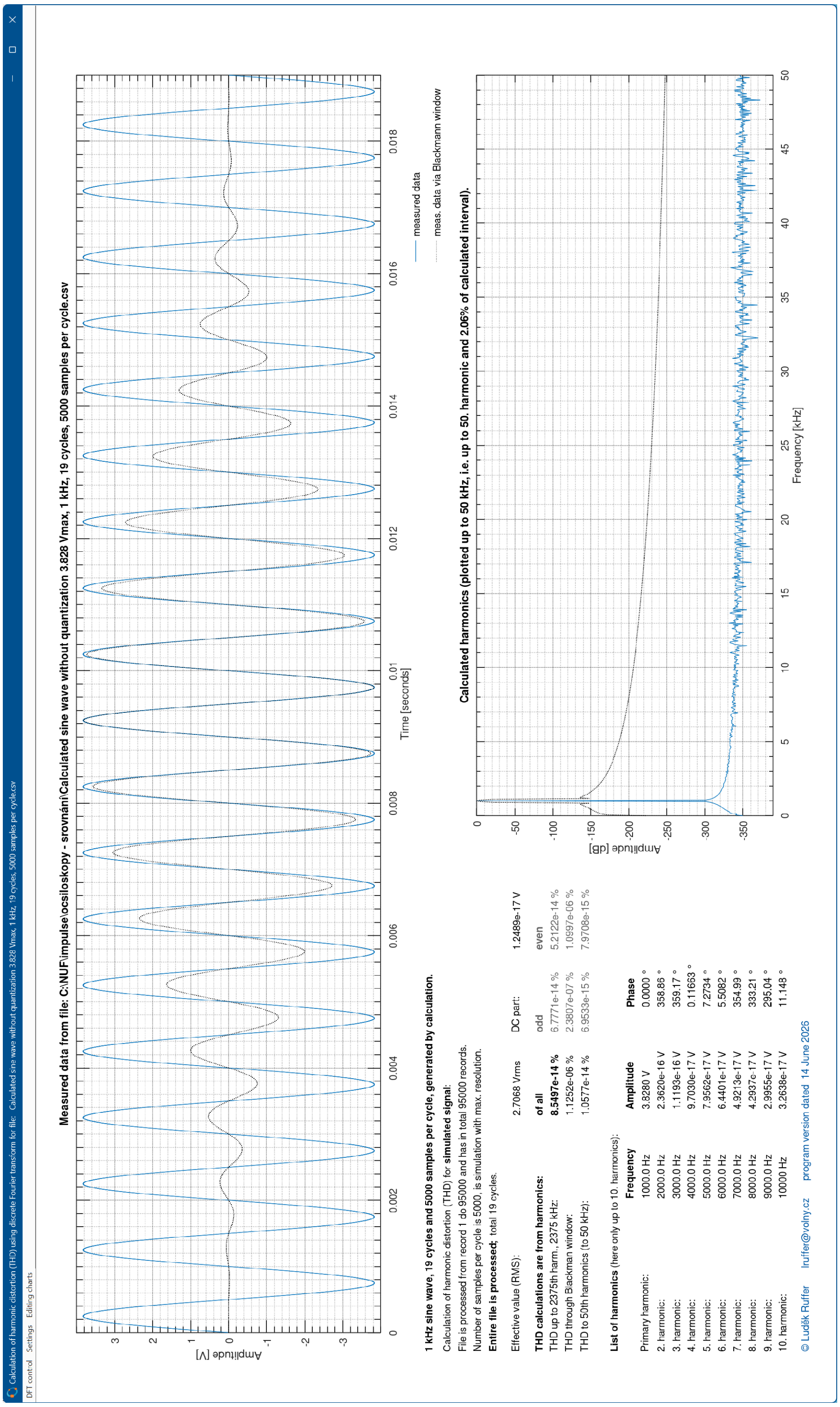
Author's notes on uncertainties of THD calculation program

Version of text from 14 July 2026

It is good custom to specify uncertainties for wheatever measuring device. And since this program processes signals measured by digital oscilloscopes, and both program and oscilloscope add uncertainties to the results, I wanted to express them here in some way. However, question of calculating THD via FFT has proven to be quite complex in terms of uncertainties, so I will at least provide descriptions of some possible errors and some sample THD calculations that can at least approximate uncertainties, since I am unable to define them precisely.



Uncertainties may, among other things, originate from FFT calculation. That is why I am inserting THD calculation on previous page for sinusoidal signal generated by calculation and without quantization in another program, using that program's maximum precision.

THD calculated in example is essentially an error, because calculated ideal sine wave cannot have any distortion, and it can be said that this false THD roughly corresponds to precision of program used to calculate sine wave (double precision). As can be seen from calculation, calculated THD is so small that it makes no sense to concern ourselves further with this uncertainty.

FFT has many attributes (according to various online sources) that can lead to incorrect results if not properly applied. These stem from:

- Discretization (sampling of an originally continuous time signal)
- From necessity to limit duration of time signal

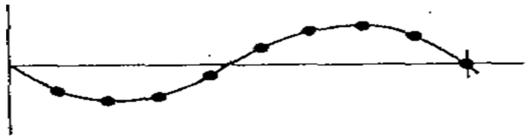
For example, the following can also lead to uncertainty:

- **Aliasing**
- **Uncertainties due to spectral leakage**
- Impact of Windows
- Filtering

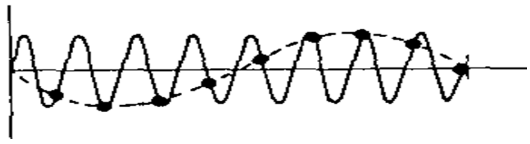
To some items in more detail:

Aliasing:
Aliasing arises from discretization (sampling) of an originally continuous time signal. It is a signal processing error caused by insufficient sampling and can corrupt digital analysis of data being examined. This occurs when a signal is sampled at a frequency too low to capture its highest frequency components, causing these high frequencies to "fold down" and appear as lower frequencies in resulting FFT spectrum. The result is spurious harmonics that never existed in actual signal, harmonics that can lead to serious misdiagnosis.

Aliasing can occur, for example, when an oscilloscope digitizes a signal and so does not record a continuous curve, but rather a sequence of discrete samples taken at fixed time intervals. If these samples are too far apart relative to the rate at which the signal changes, FFT analysis literally cannot distinguish a fast wave from a slow one. The few points it captures from the high frequency component may be combined into a perfectly plausible low frequency sine wave. This phantom low frequency is an alias, and once it appears in spectrum, it is indistinguishable from a true harmonic at that frequency. This is illustrated quite nicely in following figure:



alias – false low frequency signal



true signal with a higher frequency

To prevent this from happening, oscilloscope's sampling frequency must comply with Nyquist–Shannon sampling theorem:

To accurately represent an analog signal in digital form, sampling frequency (Fsam) must be set to at least twice highest frequency component (Fmax) present in signal (or which is to be analyzed).

This minimum sampling frequency (2 × Fmax) is called Nyquist sampling frequency. Inversely, the highest frequency that can be accurately measured at a given sampling frequency is half of that frequency: Fmax = Fsam / 2. This upper limit is Nyquist frequency. Any actual frequency above Nyquist frequency cannot be accurately expressed and will instead appear as an error in spectrum below it.

In analyzers, so called antialiasing filters are typically used to suppress aliasing. This is a steep low–pass filter that is applied to analog signal **before** it reaches analog–to–digital converter. Its purpose is to remove all frequency content above the maximum frequency (Fmax) selected by user for measurement. **It must be an analog filter. A digital filter cannot be used in this way**, especially when applied to an already digitized signal, such as Butterworth low–pass filter used in THD calculation program, because digital filter cannot remove aliasing if aliasing is already present in digitized data.

This could explain discrepancy I describe in description of program for calculating harmonic distortion in last paragraph of chapter “Modification of measured data and calculations”, namely that THD measurement of generator measured via a physical (analog, passive) low-pass filter with a cutoff frequency of approximately 25 kHz did not match digital 2nd order Butterworth filter with otherwise identical measurements, but without analog low-pass filter, yet with the same (digital) low-pass filter calculated by the program and set to roughly the same parameters.

Spectrum analyzers always include such an analog antialiasing filter, which is typically designed so that its cutoff frequency can be adjusted and is automatically set to Nyquist frequency. However, since this is obviously not an ideal filter, but has a certain slope on its descending edge, harmonics approaching Nyquist frequency are also removed. It is typically set so that harmonics are output up to approximately 80% of Nyquist frequency.

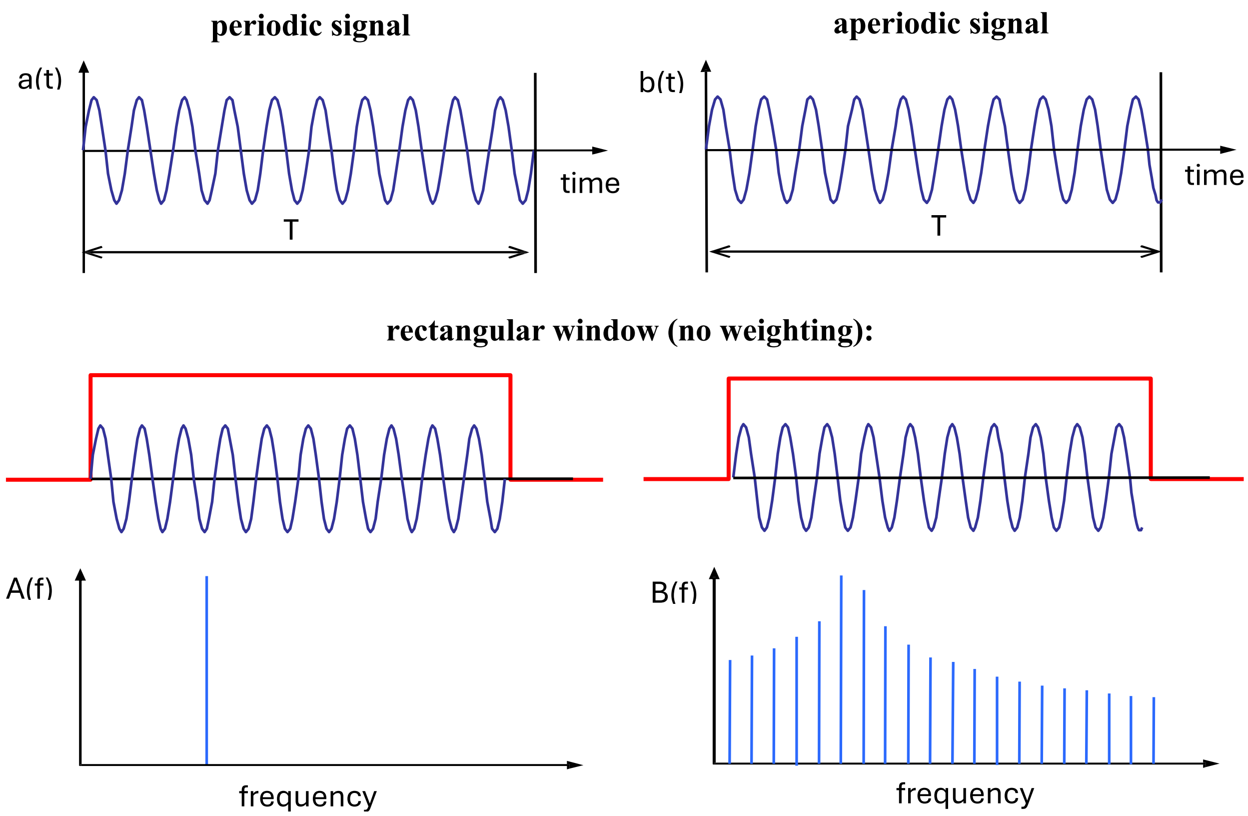
As a precaution against aliasing, the simplest step is to first verify sampling frequency to determine whether resulting Nyquist frequency is higher than highest frequency to be analyzed. According to online sources, it is best if highest frequency to be measured is no more than 80% of Nyquist frequency. If this cannot be achieved, then it will be necessary to build some kind of **analog** low–pass filter that does not have significant its own THD (i.e., passive, or at least passive in terms of the signal path, where semiconductor replacements for inductors or LC elements could be used to allow for adjustment), because oscilloscopes (that I have) do not have any such adjustable low–pass filter. However, if one wanted to use this filter for entire acoustic band, it would have to be adjustable over a fairly wide range so that it could be set to Nyquist frequencies resulting from oscilloscope sampling settings for frequencies in range of 20 Hz to 20 kHz (that is, unless one wanted to measure THD at even higher frequencies). For my oscilloscopes and settings, I use, this would range from 10 kHz (HANTEK DSO4102C oscilloscope, 20 Hz, 19 cycles recorded, Memory Depth 20k) up to 4 GHz (OWON ADS802A oscilloscope, 20 kHz, 15 cycles recorded, Acquire Depth 100k). Since this will not be feasible in practice, and I also don't know what purpose it would serve to measure, say, 20 kHz through a 4 GHz low–pass filter (in general, for low–pass filter to be that far removed from highest measured frequency), I would probably try to use a single analog low–pass filter for all measurements, set to about 50 or 100 kHz, with the caveat which would entail certain limitations: for low frequencies, it wouldn't really make a difference (I should probably test whether using a low–pass filter would be noticeable at those frequencies), and for high frequencies, it would start limiting range fairly quickly, but for frequencies in question here, that probably wouldn't be a problem (it depends on what you want to measure and whether this would be sufficient). So, I will have to give it a try; I will see.

This error will be applied more widely if Nyquist frequency of oscilloscope's sampling rate is significantly close to highest frequency to be analyzed. I hope that with today's oscilloscopes and for most frequencies in acoustic frequency bands, such low sampling frequencies are no longer encountered in digital oscilloscopes, although there are exceptions: for example, see Nyquist frequency of 10 kHz mentioned above for measurements with HANTEK oscilloscope when measuring a frequency of 20 Hz. It was encountered at memory depth of 20 k, when 20 k of data is stored in file. This oscilloscope has an even higher memory depth of 40 k, so if recording depth were set to 40 k, Nyquist frequency in this case would be double, or 20 kHz, which might be sufficient. I haven't tried comparing the two depths in this case yet, but maybe I'll give it a try; it will see.

Another way to limit influence of higher harmonics, this time those that have already been calculated, is an optional setting that determines which harmonics (by order or frequency) will be included in THD calculation, which is displayed in a one line of calculations. Here, decision regarding which harmonics to include is entirely up to program user, depending on what they expect from THD calculation.

Uncertainty due to spectral leakage:

For FFT to work properly, signal must be periodic (fundamental requirement for FFT). For this to hold true, its value at the end of a block must, among other things, be identical to its value at the beginning. If signal is not periodic, energy "leaks" into multiple harmonics close to actual frequency, and spectrum is then spread across several harmonics. Following figure illustrates this quite nicely (without using so-called windows):



This error can be fixed, or at least minimized, in following manners:

1) By changing duration of measured sample so that it matches fundamental periodicity of signal. This is program's default setting, although signal duration itself does not change (it is determined by how it is stored in oscilloscope) program attempts to select data segment that meets this requirement. In an ideal case, if were succeed to precisely select start and end of signal so that they exactly match signal's frequency, this error would be eliminated entirely. However, because oscilloscopes are sampling at a certain frequency that is not an exact multiple of measured signal's frequency, and recorded sample is not precisely bound to measured signal (it is not possible to find an exact match of start and end of period), it is maximum possible to approach ideal by a difference of one sample period. This results in a small error due to leakage, which will affect the FFT. It also depends on signal waveform, so that program can recognize when signal is rising and when it is falling; therefore, program cannot handle all waveforms, but it recognizes common, not very distorted sine waves quite well. However, it's a good idea to check this after plotting on drawn selected data waveform curve.

2) By increasing the measurement duration T will increase frequency resolution and will be finer and the effect of error will be reduced. However, with oscilloscopes, for a given frequency to be measured, this can only be achieved by reducing horizontal time base scan rate, which means that more periods will be recorded in data set. This is a bit "risky", because lowering horizontal time base scan rate results in fewer samples per oscillation, which in turn can increase error. In my experience, THD comes out better (lower), for example, for a file with approximately 20 cycles than for a file with 4 cycles. For OWON ADS802A oscilloscope, a file containing, say, 34 cycles usually

give even better results, but if there are too many cycles in file and therefore too few samples per cycle, error increases, as already mentioned here. It follows that everyone must determine optimal number of measured cycles, and thus horizontal time base scan rate, for their specific measurement on their own. I cannot specify this unequivocally, as it depends, for example, on oscilloscope used and its settings.

3) Furthermore, for example, in oscilloscopes equipped with FFT, it is common practice to “let pass” signal through mathematical filters called windows or “window transformations” before performing FFT. These windows attenuate signal to zero (or near zero) in various ways at beginning and end, while signal remains at its original amplitude in middle of waveform; this resembles a so–called bell–shaped waveform. There are several types of these windows; for comparison purposes and in case it’s not possible to select a periodic waveform, I’ve built only one window in program, namely Blackman window. THD calculated from signal through this window is always displayed in one row of calculations, and signal waveform after passing through this window is also displayed in the graph, however, since it sometimes “hid” the measured waveform, you can choose whether to display it or not.

In my experience, in most cases, THD comes out better (slightly lower) when using period selection method than when using the window method; however, as I’ve already mentioned here, it’s important to check waveform selected by program, at least by looking at it in graph, to see if the periods have been selected appropriately. Of course, choice between using period selection or window method is up to each individual.

Impact of Windows:

Functioning of windows is described in previous chapter under point 3), as for error they cause, it is also mentioned at end of previous chapter. A window (which is implemented in program) is useful in cases where it is not possible to correctly select entire periods, in such cases, THD calculated using a window may be better (lower) or even more reliable than THD calculated from a non–periodic signal. This is why windows were devised in first place: to minimize leakage error during FFT calculations and to allow FFT to be computed without need for a more complex and uncertain selection of full periods. It is way to at least perform some sort of THD calculation when no other, better option is available.

Filtering:

There are two types of filters that can be used for calculating THD:

1) **Analog filters**, which is placed before oscilloscope during measurements. They can be used, first, to suppress Aliasing, as already described in chapter on him, and second, to suppress frequencies that need to be suppressed for some reason. Thus, these can be either low–pass or high–pass filters that suppress harmonics outside their passband, which will of course affect THD, however, I assume here that if someone wants to suppress certain frequencies, it is a wanted and expected result and not an error. Just to be sure, I will reiterate here that if there is a need to limit frequency response of measured signal in any way, you **should always use analog filters connected before oscilloscope**, and they should also be passive, at least as far as signal path that signal travels through filters is concerned, so that any semiconductor components inserted

into signal path do not add their own THD to signal. Perhaps semiconductor components could be used as artificial inductances and capacitances, but they must be connected in parallel with signal. Digital filters applied to a signal that has already been digitized cannot simply affect, for example, higher frequencies that cannot be present in digitized signal due to sampling frequency; see again chapter on aliasing.

2) **Digital filters:** despite limitations described in previous paragraph, I have built Butterworth low–pass filter into program, whose parameters can be adjusted within certain range. It can be used, for example, when it is necessary to limit higher frequencies for THD calculation, which, however, must be present in digitized file, typically below approximately 80% of Nyquist frequency derived from sampling frequency, for higher frequencies, using this low–pass filter on digitized file makes no sense. Here again, I assume that if someone uses this low–pass filter, they know what they are doing and want resulting THD affected by it, so this cannot be considered an error.

That is about all I can say right now about uncertainties and methods for (sometimes partially) eliminating them. To at least provide some description of errors that may occur in various measurements, I've included tables below summarizing THD calculated from the same waveform generated by the same signal generator, first for different signal amplitudes relative to the same vertical range of oscilloscope, second, for different numbers of signal periods, and third, for waveforms captured by two oscilloscopes with 8bit and 12bit AD converters. The reason for this is so that I could somehow demonstrate differences in THD that arise from these methods, even though input signal remains the same. However, on following page first presents comparison table for signal calculated using different program.

Comparison of ideal (calculated) signals for 8bit and 12bit quantization and for different signal amplitudes (relative to quantization level).

Calculated 1 kHz sine wave quantized to 8 bits (as 8bit A/D converter) quantization was always based on total 8V range, as for sensitivity of 1 V/div. Calculated THD for various input voltages and numbers of processed cycles.				**ϕ THD (it clearly does not depend on number of periods)**
Input voltage Uin / range	Number of periods processed / samples per period			
	9 / 2000	19 / 1000	39 / 500	
3,828 Vmax	0,34064 %	0,34314 %	0,34152 %	0,3418 %
0,9570	-49,35 dB	-49,29 dB	-49,33 dB	-49,33 dB
1,914 Vmax	0,68450 %	0,69064 %	0,69197 %	0,6890 %
0,4785	-43,29 dB	-43,21 dB	-43,20 dB	-43,24 dB
0,957 Vmax	1,29550 %	1,30330 %	1,30110 %	1,3000 %
0,2393	-37,75 dB	-37,70 dB	-37,71 dB	-37,72 dB

Calculated 1 kHz sine wave quantized to 12 bits (as 12bit A/D converter) quantization was always based on total 8V range, as for sensitivity of 1 V/div. Calculated THD for various input voltages and numbers of processed cycles.				**ϕ THD (it clearly does not depend on number of periods)**
Input voltage Uin / range	Number of periods processed / samples per period			
	3 / 25000	19 / 5000	39 / 2500	
3,828 Vmax	0,01914 %	0,02247 %	0,02276 %	0,0215 %
0,9570	-74,36 dB	-72,97 dB	-72,86 dB	-73,37 dB
1,914 Vmax	0,04238 %	0,04420 %	0,04423 %	0,0436 %
0,4785	-67,46 dB	-67,09 dB	-67,08 dB	-67,21 dB
0,957 Vmax	0,08075 %	0,08003 %	0,08182 %	0,0809 %
0,2393	-61,86 dB	-61,93 dB	-61,74 dB	-61,84 dB

Here, it can be seen that in this case, number of processed cycles (periods) has almost no effect on result, as the signal is generated by a calculation and is therefore completely regular and identical across all periods. On the other hand, magnitude of processed voltage relative to oscilloscope’s vertical range has a significant effect. This is entirely logical and is always the case with any measuring instrument. Description of program for calculating THD states that THD of device under test must be measured twice: first, THD of entire chain (generator + device) is measured, and second, under the same conditions, only THD of generator is measured, and resulting THD is calculated as difference between the two. Therefore, when calculating THD with this program, it follows that it is absolutely essential for magnitude of sinusoidal voltage to be the same for measuring of both, entire chain and generator; otherwise, difference would be reflected as error. However, this cannot always be ensured, for example, when measuring THD of a power amplifier, since its voltage at full power will be higher than generator’s maximum voltage. This can be mitigated, at least to some extent, if **will be to measure generator’s THD at voltage that will be in the same ratio as generator’s voltage to smaller range of oscilloscope used for its measurement, as will be amplifier’s voltage at full power to higher range of oscilloscope**. However, error will never be exactly the same as it would be with identical ranges, but it’s at least something.

A comparison of THD values calculated from measurements of the same signal taken with oscilloscopes equipped with 8bit and 12bit AD converters at their various ranges and for different signal amplitudes (relative to the oscilloscope's range).

HANTEK DSO4102C oscilloscope (8bit AD converter), 1 kHz sine wave from OWON DGE1030*) generator; vertical range was 1 V/div throughout. Calculated THD for various input voltages and numbers of processed cycles.

Input voltage	Number of periods processed / samples per period		
Uin / range **)	9 / 2000	19 / 1000	39 / 500
3,828 Vmax	0,40303 %	0,30341 %	0,23144 %
0,9570	-47,89 dB	-50,36 dB	-52,71 dB
1,914 Vmax	0,83998 %	0,64621 %	0,52194 %
0,4785	-41,51 dB	-43,79 dB	-45,65 dB
0,957 Vmax	1,53800 %	1,34790 %	1,16140 %
0,2393	-36,26 dB	-37,41 dB	-38,70 dB

HANTEK DSO4102C oscilloscope (8bit AD converter), 1 kHz sine wave from OWON DGE1030*) generator; vertical range was 0.5 V/div throughout. Calculated THD for various input voltages and numbers of processed cycles.

Input voltage	Number of periods processed / samples per period		
Uin / range **)	9 / 2000	19 / 1000	39 / 500
1,914 Vmax	0,45639 %	0,34530 %	0,28100 %
0,9570	-46,81 dB	-49,24 dB	-51,03 dB
0,957 Vmax	0,97791 %	0,79604 %	0,68760 %
0,4785	-40,19 dB	-41,98 dB	-43,25 dB
0,479 Vmax	1,92860 %	1,57160 %	1,36540 %
0,2395	-34,30 dB	-36,07 dB	-37,29 dB

OWON ADS802A oscilloscope (12bit AD converter), 1 kHz sine wave from OWON DGE1030*) generator; vertical range was 1 V/div throughout. Calculated THD for various input voltages and numbers of processed cycles.

Input voltage	Number of periods processed / samples per period		
Uin / range **)	3 / 25000	19 / 5000	39 / 2500
3,828 Vmax	0,14219 %	0,07177 %	0,05944 %
0,9570	-56,94 dB	-62,88 dB	-64,52 dB
1,914 Vmax	0,26920 %	0,12720 %	0,09361 %
0,4785	-51,40 dB	-57,91 dB	-60,57 dB
0,957 Vmax	0,49609 %	0,22362 %	0,18088 %
0,2393	-46,09 dB	-53,01 dB	-54,85 dB

OWON ADS802A oscilloscope (12bit AD converter), 1 kHz sine wave from OWON DGE1030*) generator; vertical range was 0.5 V/div throughout. Calculated THD for various input voltages and numbers of processed cycles.

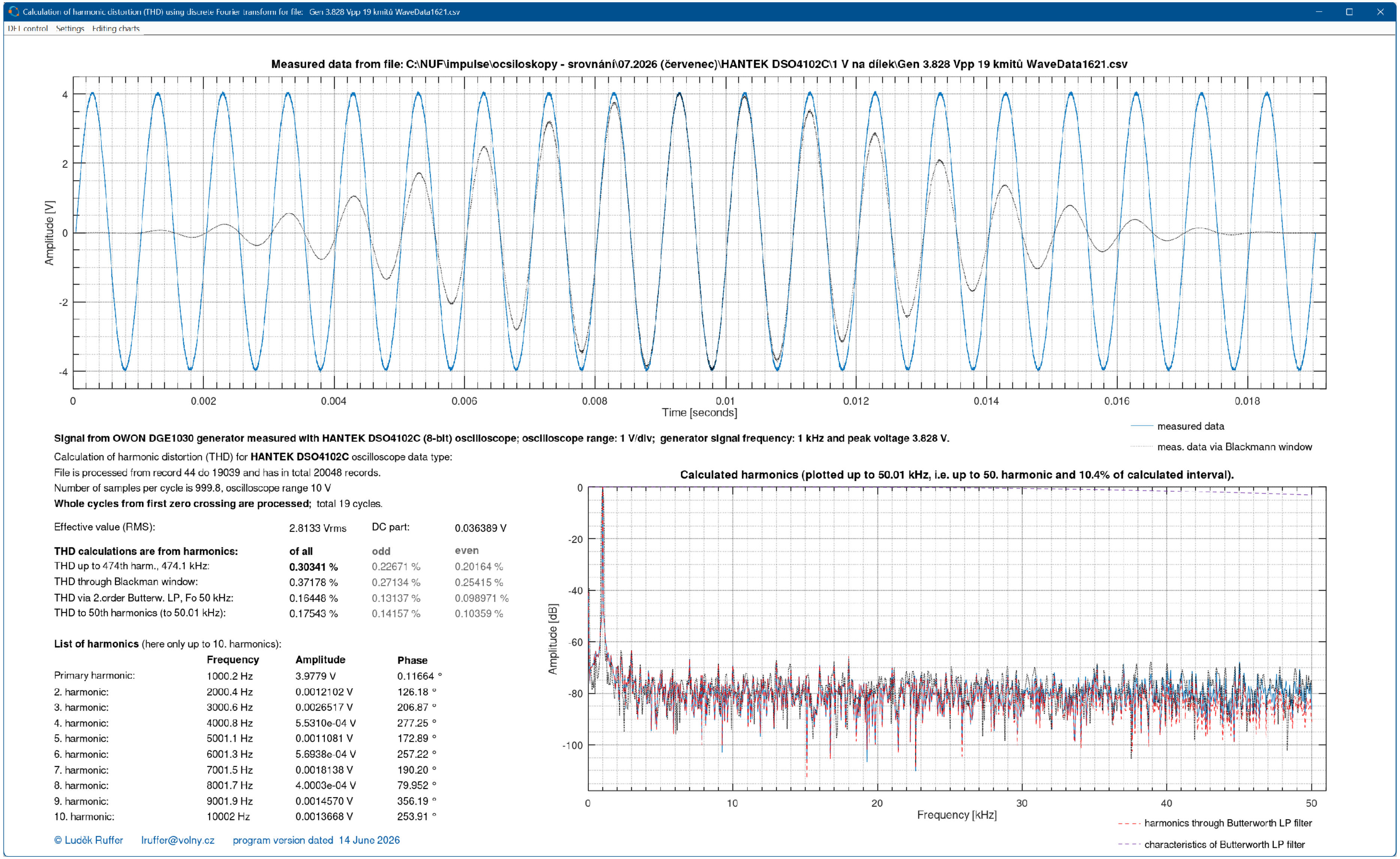
Input voltage	Number of periods processed / samples per period		
Uin / range **)	3 / 25000	19 / 5000	39 / 2500
1,914 Vmax	0,14197 %	0,07584 %	0,06012 %
0,9570	-56,96 dB	-62,40 dB	-64,42 dB
0,957 Vmax	0,28498 %	0,16487 %	0,11066 %
0,4785	-50,90 dB	-55,66 dB	-59,12 dB
0,479 Vmax	0,55934 %	0,29088 %	0,23649 %
0,2395	-45,05 dB	-50,73 dB	-52,52 dB

*) Note: harmonic distortion of the generator's signal is defined twice in its documentation:
1 - Harmonic distortion: Typically (0dBm) for DC to 1 MHz: <-65dBc (that is <0,05623..%)
2 - Total harmonic distortion: <0.2%, 10 Hz to 20 kHz; 1 Vpp

**) Ratio Uin / range is calculated for peak-to-peak input voltage, i.e. Vmax * 2
and range is determined by setting sensitivity in V/division and applies across entire screen, which has 8 divisions on both oscilloscopes, i.e. (V/division) * 8

From this table, it can be seen that for real signal measured with oscilloscope, number of periods processed is also important for calculating THD – up to a certain limit, more periods there are, lower the THD. This is likely because individual waves here are not exactly the same as in the previous case, but are slightly “shaggy”. **Therefore, when measuring THD of real signals, it is also necessary to maintain the same number of cycles at the same sampling frequency for measuring THD of both entire signal chain and generator.**

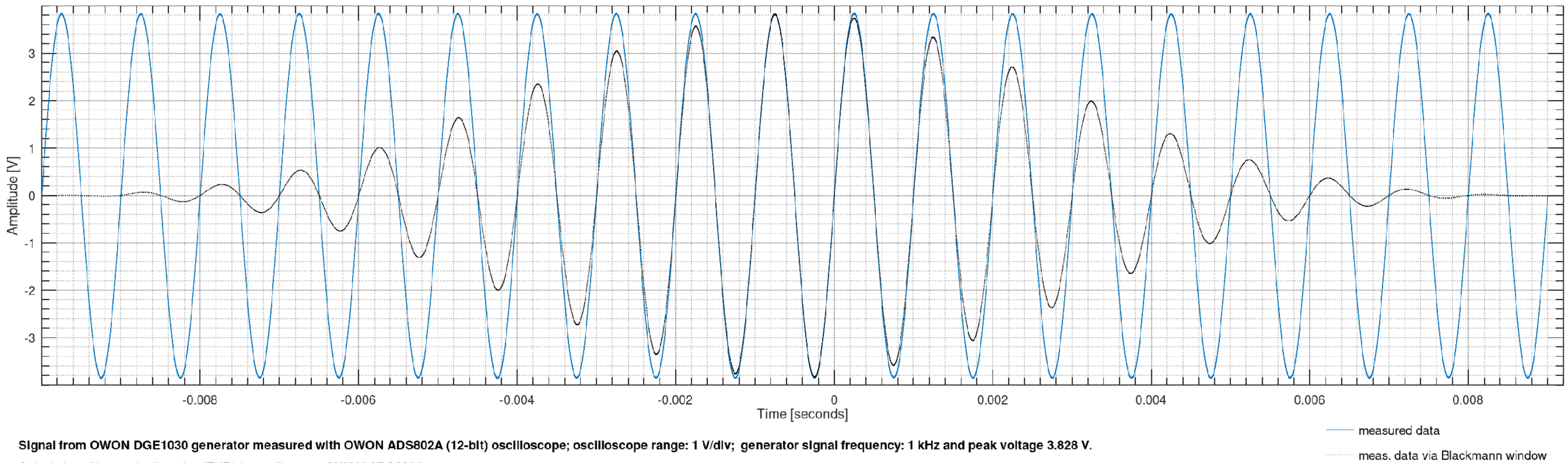
Here are some examples of THD calculations from the previous table.



Calculation of harmonic distortion (THD) using discrete Fourier transform for file: Gen 3.828 Vpp 19 kmitů 20260707152524.csv

DFT control Settings Editing charts

Measured data from file: C:\NUF\impulse\ocsiloskopy - srovnání\07.2026 (červenec)\OWON ADS802A\1 V na dílek\Gen 3.828 Vpp 19 kmitů_20260707152524.csv



Signal from OWON DGE1030 generator measured with OWON ADS802A (12-bit) oscilloscope; oscilloscope range: 1 V/div; generator signal frequency: 1 kHz and peak voltage 3.828 V.

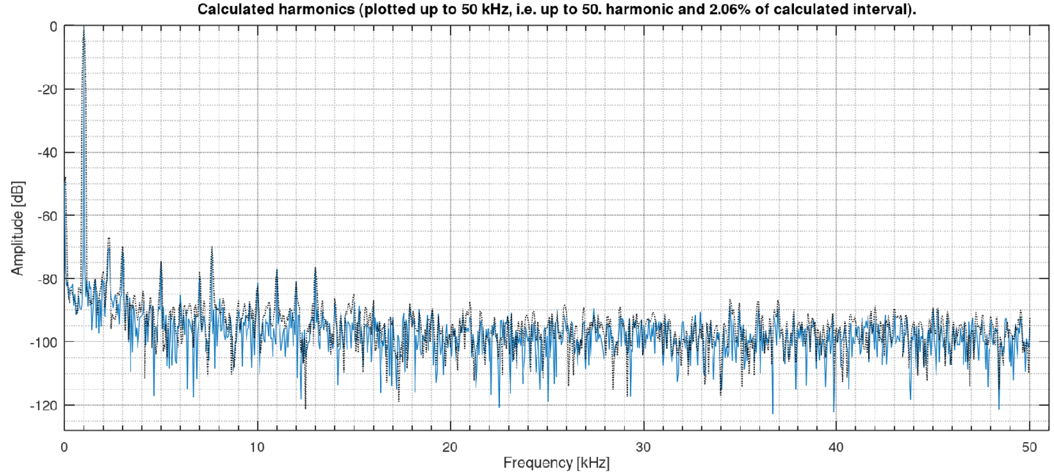
Calculation of harmonic distortion (THD) for oscilloscope **OWON ADS802A**:
File is processed from record 14 do 95012 and has in total 100000 records.
Number of samples per cycle is 5000, oscilloscope range 8 V, channel CH1
Whole cycles from first zero crossing are processed; total 19 cycles.

Effective value (RMS):	2.7151 Vrms	DC part:	0.013364 V
THD calculations are from harmonics:	**of all**	odd	even
THD up to 2374th harm., 2374 kHz:	**0.071765 %**	0.057341 %	0.043154 %
THD through Blackman window:	0.089591 %	0.070808 %	0.054889 %
THD via 2.order Butterw. LP, Fo 50 kHz:	0.044108 %	0.040016 %	0.018553 %
THD to 50th harmonics (to 50 kHz):	0.043949 %	0.039805 %	0.018630 %

List of harmonics (here only up to 10. harmonics):

	Frequency	**Amplitude**	**Phase**
Primary harmonic:	1000.0 Hz	3.8396 V	0.28581 °
2. harmonic:	2000.0 Hz	3.4439e-04 V	312.01 °
3. harmonic:	3000.0 Hz	9.9624e-04 V	189.22 °
4. harmonic:	4000.0 Hz	9.8064e-05 V	221.12 °
5. harmonic:	5000.1 Hz	7.1168e-04 V	187.31 °
6. harmonic:	6000.1 Hz	2.0962e-04 V	250.51 °
7. harmonic:	7000.1 Hz	3.8860e-04 V	17.682 °
8. harmonic:	8000.1 Hz	1.1877e-04 V	84.261 °
9. harmonic:	9000.1 Hz	1.1695e-04 V	182.87 °
10. harmonic:	10000 Hz	3.2251e-04 V	69.250 °

Calculated harmonics (plotted up to 50 kHz, i.e. up to 50. harmonic and 2.06% of calculated interval).



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Well, both calculations show that AD converter of oscilloscope used also has a significant impact on THD (i.e., its error); in my case, this means whether signal is quantized with 8–bit or 12–bit resolution, which is, after all, logical and predictable. Eight-bit AD converter divides signal into 256 levels (2^8) from oscilloscope's range; for vertical range across entire screen, e.g., 8 V, "steps" will be 8 / 256 = 0.03125 V, meaning "dead zone" of approximately 0.3906% of whole range. Twelve-bit AD converter divides signal into 4096 levels (2^{12}) from oscilloscope's used range; for vertical range spanning entire screen, e.g., 8 V, "steps" will be 8 / 4096 = 0.001953125 V, resulting in "dead band" of approximately 0.02441% of whole range. An AD converter with a resolution even higher than 12 bits mentioned here will, of course, have an even smaller error.

Tables with measures of two channels of power amplifier at power approx. 10.4 W and 7.3 W:

Measurements of amplifier from czech *Radiový konstruktér* 5/1974, p. 17, Fig. 17, in box, finished on 27 June 2026, with only EMI filter												
F [Hz]	Amplifier 1 (on channel 1)				Amplifier 2 (on channel 2)				Generator on oscilloscope CH. 1		Generator on oscilloscope CH. 2	
	Urms [V]	THD amplifier + generator [%]	**THD amplifier [%]**	Characteristics [dB]	Urms [V]	THD amplifier + generator [%]	**THD amplifier [%]**	Characteristics [dB]	Urms [V]	THD [%]	Urms [V]	THD [%]
20	6,3693	0,16701	**0,0910**	-0,155	6,4063	0,18893	**0,1163**	-0,133	2,7059	0,076015	2,7125	0,072602
25	6,4294	0,11993	**0,0468**	-0,073	6,4593	0,10008	**0,0299**	-0,062	2,7082	0,073136	2,7149	0,070146
50	6,5263	0,11222	**0,0346**	+0,056	6,5471	0,11161	**0,0386**	+0,055	2,7192	0,077611	2,7260	0,073053
100	6,5345	0,11544	**0,0242**	+0,067	6,5530	0,11228	**0,0383**	+0,063	2,7133	0,091248	2,7221	0,073968
500	6,4855	0,11472	**0,0383**	+0,002	6,5086	0,10961	**0,0336**	+0,004	2,6899	0,076420	2,7010	0,075972
1000	6,4840	0,11322	**0,0289**	0	6,5055	0,12366	**0,0465**	0	2,6882	0,084274	2,6991	0,077202
5000	6,4600	0,15391	**0,0609**	-0,032	6,4844	0,15303	**0,0792**	-0,028	2,6830	0,093057	2,6942	0,073804
10000	6,4483	0,22998	**0,1508**	-0,048	6,4796	0,29007	**0,2153**	-0,035	2,6822	0,079216	2,6938	0,074809
15000	6,4314	0,38537	**0,2976**	-0,071	6,4609	0,49371	**0,4196**	-0,060	2,6832	0,087819	2,6939	0,074119
20000	6,4133	0,59650	**0,5251**	-0,095	6,4409	0,73567	**0,6566**	-0,087	2,6826	0,071419	2,6943	0,079112
25000	6,3941	1,22780	1,1345	-0,121	6,4229	0,99598	0,9211	-0,111	2,6838	0,093280	2,6928	0,074893
	Amplifier 1 (on channel 1) in range of 20 Hz to 20 kHz						**Amplifier 2 (on channel 2) in range of 20 Hz to 20 kHz**					
	THD ϕ :		**0,130 %**	Δ Charact. :	+0,067 dB	-0,155 dB	**THD ϕ :**		**0,167 %**	Δ Charact. :	+0,063 dB	-0,133 dB
	Load resistor:	4,0 Ω	Urms ϕ :	6,458 V	Power ϕ :	10,427 W	Load resistor:	4,0 Ω	Urms ϕ :	6,485 V	Power ϕ :	10,512 W

Measurements of amplifier from czech *Radiový konstruktér* 5/1974, p. 17, Fig. 17, in box, finished on 27 June 2026, with only EMI filter												
F [Hz]	Amplifier 1 (on channel 1)				Amplifier 2 (on channel 2)				Generator on oscilloscope CH. 1		Generator on oscilloscope CH. 2	
	Urms [V]	THD amplifier + generator [%]	**THD amplifier [%]**	Characteristics [dB]	Urms [V]	THD amplifier + generator [%]	**THD amplifier [%]**	Characteristics [dB]	Urms [V]	THD [%]	Urms [V]	THD [%]
20	5,3890	0,09904	**0,0230**	-0,149	5,3584	0,08072	**0,0081**	-0,128	2,7059	0,076015	2,7125	0,072602
25	5,4378	0,08461	**0,0115**	-0,071	5,3983	0,07450	**0,0044**	-0,063	2,7082	0,073136	2,7149	0,070146
50	5,5211	0,08544	**0,0078**	+0,061	5,4707	0,06519	**-0,0079**	+0,052	2,7192	0,077611	2,726	0,073053
100	5,5248	0,07743	**-0,0138**	+0,067	5,4785	0,07175	**-0,0022**	+0,065	2,7133	0,091248	2,7221	0,073968
500	5,4855	0,07908	**0,0027**	+0,005	5,4441	0,07061	**-0,0054**	+0,010	2,6899	0,076420	2,701	0,075972
1000	5,4822	0,07765	**-0,0066**	0	5,4378	0,07343	**-0,0038**	0	2,6882	0,084274	2,6991	0,077202
5000	5,4612	0,09544	**0,0024**	-0,033	5,4231	0,11460	**0,0408**	-0,024	2,683	0,093057	2,6942	0,073804
10000	5,4450	0,16051	**0,0813**	-0,059	5,4101	0,24640	**0,1716**	-0,044	2,6822	0,079216	2,6938	0,074809
15000	5,4252	0,24867	**0,1609**	-0,091	5,3911	0,42424	**0,3501**	-0,075	2,6832	0,087819	2,6939	0,074119
20000	5,3982	0,42139	**0,3500**	-0,134	5,3640	0,64149	**0,5624**	-0,119	2,6826	0,071419	2,6943	0,079112
25000	5,3616	0,72258	0,6293	-0,193	5,3283	0,84154	0,7666	-0,177	2,6838	0,093280	2,6928	0,074893
	Amplifier 1 (on channel 1) in range of 20 Hz to 20 kHz						**Amplifier 2 (on channel 2) in range of 20 Hz to 20 kHz**					
	THD ϕ :		**0,062 %**	Δ Charact. :	+0,067 dB	-0,149 dB	**THD ϕ :**		**0,112 %**	Δ Charackt. :	+0,065 dB	-0,128 dB
	Load resistor:	4,0 Ω	Urms ϕ :	5,457 V	Power ϕ :	7,445 W	Load resistor:	4,0 Ω	Urms ϕ :	5,418 V	Power ϕ :	7,338 W

These Excel spreadsheets contain two sets of measurements and calculate, for each measured point (frequency) and measured channel, THD as difference between THD of entire chain and THD of generator itself, which was used to measure amplifier. Oscilloscope used was a 12–bit OWON ADS802A, and generator was an OWON DGE1030, which, according to its documentation, is supposed to have a harmonic distortion of "*Typically (0 dBm) for DC to 1 MHz: <–65 dBc*" (i.e., <0.05623…%), but elsewhere in documentation it states that it has a vertical resolution of 14 bits, which would imply an even lower THD. However, it is necessary to adhere to distortion values specified in the documentation, and I assume they have their reasons for doing so. However, I would strongly recommend that signal generator used to measure THD in conjunction with a digital oscilloscope have a higher vertical resolution than oscilloscope itself, as in this case.

As can be seen from each table, calculated THD can sometimes be slightly negative, which may seem like nonsense, since it can't actually be negative, but in reality, this says something about the resolution of this method. It shows that resolution (in the case of oscilloscopes with a 12-bit AD converter) is approximately 0.02% THD, and to be on the safe side, I'll assume it's better than **0.05% THD**. I don't think that's such a bad result, considering the means by which this value has been achieved—oscilloscope can also function as an oscilloscope, so it can be considered multipurpose. And I'm not even talking about the price of professional analyzers for audio band. For example, I saw used Keysight U8903A analyzer (10 Hz to 100 kHz) for sale on eBay for US $5200… and while it's better in terms of accuracy by an order of magnitude, my OWON ADS802A oscilloscope, along with OWON DGE1030 generator, worked out to be significantly better value for money. In fall of 2025, I paid converted to US dollars of about US $400 for both of them combined, including shipping, VAT, and fees; see, for example, ADS802A oscilloscope (if it's still listed there).

For sake of completeness, it would also be necessary to account for uncertainty of vertical amplifier; for example, for my oscilloscopes, documentation for OWON ADS802A oscilloscope states, "*DC gain accuracy (for) ≥5 mV is 1.5%*" and for HANTEK DSO4102C oscilloscope, documentation states, "*DC gain accuracy: ±3% for Normal or Average acquisition mode, 5V/div to 10mV/div*"; however, documentation for neither oscilloscope explains what these percentages are based on… This uncertainty affects magnitude of measured signal amplitude and, consequently, magnitudes of the calculated harmonics.

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