Fast Distortion Measurements in Relation to Frequency and Level

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Introduction

To obtain non-linear system characteristics, several methods for measuring parameters such as THD and IMD are described in the literature. With these measurement techniques, a device under test (DUT) can be measured to characterize its non-linear behaviour. However, most of these techniques are quite slow, which not only results in long measurement times, but also leads to side effects (e.g. a heating of the voice coil in loudspeakers).

It is therefore important to deploy fast measurement techniques, which allow a rapid evaluation of non-linear system parameters with sufficient accuracy. These can be useful for permanent quality controls in a production line or as an evaluation tool during development.

The accuracy of the proposed methods should be variable, but at least equal to the accuracy of the classic methods. The variability allows even faster measurements if only a rough estimation is needed.

In this paper only the total harmonic distortion (THD) is discussed. The THD is normally given by the sum of energy of each harmonic that falls into the audible range (up to 20 kHz) in relation to the signal itself. Although the numerator is well defined there exists some confusion about the denominator. What exactly is "the signal"? The traditional definition describes the signal as RMS value of the fundamental and the harmonics. However, in ISO 60268-2 and in AES17 the noise is considered as well in the nominator. This has the advantage that the distortion can never exceed 100%. However, in normal conditions (good SNR) both definitions result in nearly equal THD values.

Pure Tone Distortion Measurements

The traditional method to determine the THD is based on single measurements of a pure tone at a fixed level and frequency. Therefore an ultra-low noise sine generator excites the DUT. In case of coherent sampling, the harmonics can be obtained by an FFT of the output signal. To achieve coherent sampling, the excitation signal frequency must be chosen to match exactly with one of the frequency bins of the FFT spectrum. It then is an integer multiple of the bin distance *df*, which depends on the sampling rate f_S and the FFT-order N:

$$df = \frac{f_s}{2^N} \qquad f_{sig} = k \cdot df \qquad k = 1, 2, 3, 2^N$$

Coherent sampling guarantees that an integer number of signal periods will fall into the FFT interval. This allows an error-free FFT analysis with the energy of each component concentrated in just one bin.

If no coherent sampling is possible, window functions must be used in order to suppress errors introduced by the discontinuity at the transition between the last and the first sample of the FFT. This method is called non-coherent sampling.

In this case, it is necessary to not only consider the actual bin (either of the fundamental or the harmonics), but the sum of bins that falls into the main lobe. Although theoretically the side lobes must be considered as well, it is better to only consider the main lobe, since the side lobes contribute very little energy, but worsen the signal-to-noise ratio.

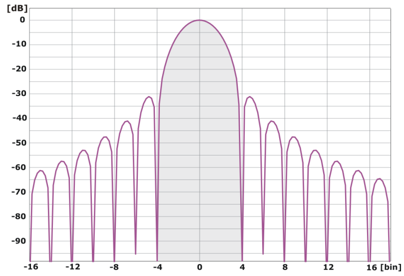

Figure 1: Spectral domain view of a hamming window. Gray: area considered for THD calculation

By choosing a window function with a good side lobe suppression such as the "Albrecht"-Windows, these problems can be minimized.

Distortion vs. Frequency

For most technical applications the THD is only calculated at 1 kHz. However, in some cases it is also necessary to see how the distortion develops over different frequencies. Therefore several pure tone measurements at different frequencies are performed. Usually the frequencies can either be spaced with a fixed absolute distance (e.g. 100 Hz, linear spaced) or with a fixed relative ratio (e.g. 1/12 octave, logarithmically spaced). To ensure coherent sampling it might be necessary to round the frequencies to fit into the given FFT block length.

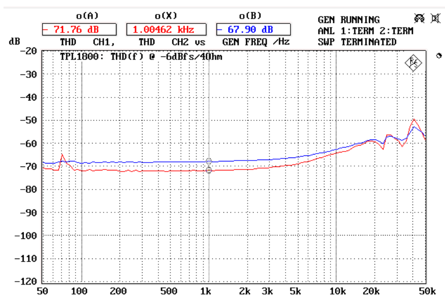

Figure 2: Distortion vs. Frequency for a low quality AD converter.

Between each measurement it is necessary to wait until the new frequency is in a stable condition (i.e. out of transient oscillation caused by a filter). This results in a longer measurement duration, which can lead to unwanted side effects.

Simultaneous Measurement of Transfer Function and Harmonic Distortion

To obtain the complex response of a system it is common to use a sine sweep as excitation signal. In contrast to other measurement techniques such as MLS it has the advantage that distortion artefacts can be windowed out. But not only this. In case of a so called 'logarithmic sweep' (i.e. a sine sweep which increases the frequency in octaves per second) an interesting phenomenon can be seen: After deconvolution, for each harmonic order a dedicated impulse response (IR) can be found at a specific position prior to the fundamental impulse response. These can be separated well from the impulse response and isolated with the use of window functions. [1]

To achieve this goal, the excitation signal should be considerably longer than the IR. Alternatively, zeros can be inserted to perform a non-periodic deconvolution. The second half of the IR interval can then be interpreted as negative time.

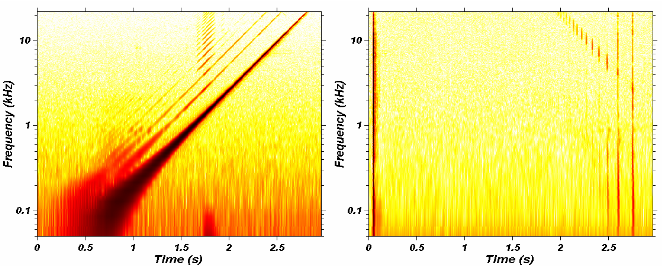

Figure 3: Time-Frequency-Plot of a non-linear device excited with a 'Log Sweep', before and after the deconvolution.

By relating the harmonic impulse responses to the fundamental, the frequency dependent distortion can be evaluated. The method has been published first by A. Farina [2]. It can be shown that even other warp functions of the group delay can be used as well to generate a sweep (e.g. a linear sweep). However, in such a case it would be necessary to use a separate reference spectrum for each harmonic to stake them to straight lines (see figure 3). Furthermore, the distance between the components of the individual harmonics would not be frequency-independent any more.

After deconvolution with the reference spectrum, the IR can be obtained by an IFFT of the spectrum. The actual IR can now be found at the left border and a couple of “harmonic impulse responses” (HIRs) at negative times near the right border. The distance between the HIR can be calculated by:

$$Dist_{HIR\,AB} = \frac{\log_2(ord_A / ord_B)}{sweep\,rate\,[oct/s]}$$

The fundamental and each HIR can now be separated by windowing. Each obtained part is usually much shorter than the whole measurement size, thus for further analysing, each chunk can be transformed by an FFT with a much shorter block length than the one used for the initial deconvolution of the sweep response. This speeds up the processing, especially if dozens of HIRs need to be evaluated. To relate the frequency contents of one HIR spectrum to the fundamental, a spectral shift operation according to the order of the specific harmonic must be performed. For example, in case of the fourth-order harmonic a shift to one-fourth of the original frequency must be applied. After this operation, the shifted spectrum can be divided by the fundamental spectrum, yielding the frequency-dependent distortion fraction.

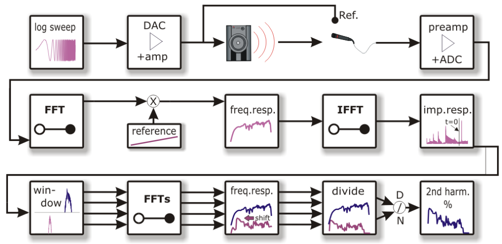

Figure 4: Logarithmic frequency sweep sequence chart

In comparison to the standard single-tone excitation and analysis with fixed frequency increments, the 'Log Sweep' method is many times faster. Furthermore, the frequency resolution is much higher in the high-frequency and usually also still higher in the mid frequency range, depending on the FFT size chosen. Also a high FFT order must be chosen, if too much reverberation leads to smearing of the distinct HIRs into each other by delayed components. This restriction generally limits this method to be performed only under anechoic conditions.

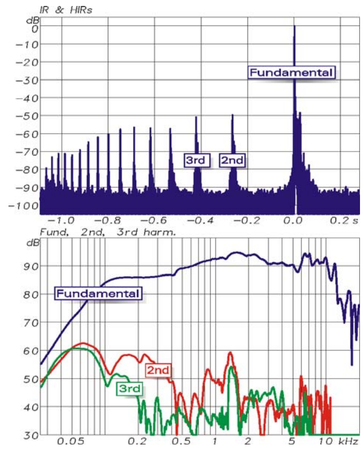

Figure 5: IR of the fundamental and its harmonics (right to left) and the corresponding spectra (2^{nd} and 3^{rd} order)

Distortion vs. Level

Besides measuring the THD over the frequency, it is also of interest to measure the THD over the level. Again, this can be performed either logarithmically with levels increasing in dB or linearly with an increasing amplitude in fixed steps.

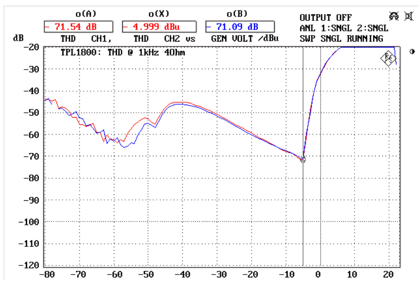

Figure 6: Distortion vs. Level for a low quality DA converter.

Fast Distortion vs. Level Measurement

Similar to the above described fast measurement technique for distortion vs. frequency a significant speed-up in measurement time can be gained by sweeping the level. Especially since the necessary settle time at the beginning of each excitation signal emission slows down the overall procedure. The measurement technique described here was proposed by S. Müller as the so called 'Level Sweeps' [3].

The idea behind this technique is to continuously increase the level rather then stepping through with fixed amplitude steps. After applying an inverse fade function to the retrieved signal, several FFT snapshots are taken during the sweep, keeping track of the THD at the current level. To avoid the discontinuities between the last and the first sample of each snapshot, a reverse fade factor must be applied to the recorded level sweep to achieve a more or less constant amplitude. The remaining variations are caused by the level dependent non-linearity. If those effects are too large, window functions must be used to window out a smaller area. Again all the usual problems associated with windowing apply here.

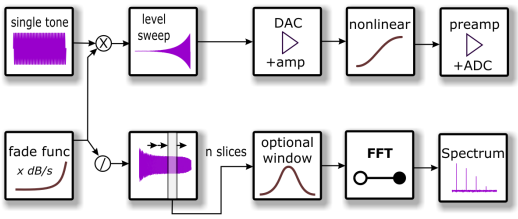

Figure 7: Level sweep sequence chart

For each snapshot, the necessary parameters for calculating the THD can be extracted and plotted. This results in a THD over level diagram. The resolution of levels is only limited by the computing power and not necessarily by the measurement duration. With all the enhancements described in the first chapter, several hundred FFT snapshots are possible in real-time, allowing a resolution detailed enough for all needs.

Despite all the advantages several limitations exist. First of all it is important to choose an adequate FFT order. As holds for steady state measurements, a small FFT order allows for very fast testing, while a higher order improves spectral selectivity and decreases the relative height of the spectral bins containing uncorrelated noise.

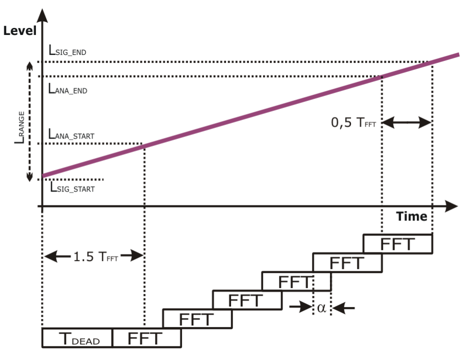

Figure 8: Relationship between start and end levels of the excitation signal and the analysis.

For coherent sampling, the signal must fit exactly into an FFT size, which means that within this size the signal has an integer period. By choosing a higher FFT order, a finer grid is available at the expense of a higher CPU usage. Furthermore, the FFT order limits the maximum level measurable near 0 dB FS (dB relative to full-scale).

To shape the sine wave in the means of level, different ramp functions can be considered. The most practical and easiest to implement is the exponential progression, leading to a constant level increment per time dL/dt in dB/s.

The total length of the excitation signal is a function of the level range L_{RANGE}, desired analysis level increment ΔL, FFT-order N and overlap factor α (0: no overlap, 1: full overlap) between each FFT segment:

$$T_{sig} = T_{FFT} \cdot \left(1 + \frac{L_{RANGE}}{\Delta L}(1-\alpha)\right) \quad \text{with} \quad T_{FFT} = \frac{2^N}{f_s}$$

The level range L_{RANGE} of the excitation signal must be somewhat larger than the intended analysis range. Prior to the start, an additional dead time is required in order to eliminate artefacts originating from the sudden onset of the sine wave. In accordance to the classical stepped THD measurement a typical value for T_{DEAD} is the length T_{FFT} of an FFT interval. To calculate the signal start level, an additional half FFT interval must be considered. Thus the start level can be calculated as:

$$L_{sig.start} = L_{ana.start} - 1.5 \cdot T_{FFT} \cdot \frac{\delta L}{\delta t}$$

Likewise, half an FFT interval must be added to the analysis end level to obtain the excitation signal's end level:

$$L_{sig.end} = L_{ana.end} + 0.5 \cdot T_{FFT} \cdot \frac{\delta L}{\delta t}$$

In case the DUT is the DAC itself, the signal end level $L_{sig.end}$ cannot rise above 0 dB FS. This means that the analysis can only be performed up to $L_{ana.end}$, which lies $0.5 \cdot T_{FFT} \cdot \frac{\delta L}{\delta t}$ (half the level swept through in one FFT interval) below 0 dB FS. For measuring up to 0 dB FS, at least one measurement must be performed with the traditional method. Furthermore, additional discrete level measurements are necessary if the overlap factor α is higher than 50%.

After the level sweep response has been captured, an inverse fade function must be applied to level-equalize the signal. As it holds for perfectly linear DUTs, this will restore the excitation signal entirely and result in a constant envelope. In contrast, a DUT with strong non-linearities will lead to a deviation from the ideal constant envelope. Especially when viewed on a logarithmic amplitude scale with a small dB range, the distortion artefacts can be detected visually.

However, even after level-equalization a certain amount of discontinuity between the edges of each individual FFT segment can be found. This is because the amount of level reduction and the accompanying distortion are level-dependent.

The level-equalized DUT output signal is analysed in narrow segments of 2^N samples. The segments are taken from left to right, with their start point incrementing according to the desired level resolution ΔL :

$$\Delta t = \frac{\Delta}{\delta L / \delta t}$$

To increase the resolution, the slices can overlap each other. Especially when using windowed FFT analysis, at least 50% of overlap is recommended to avoid loosing details near the edges of each segment.

To extract the distortion components, an FFT is performed for each segment. After component separation all different types of calculation can be used without measuring the DUT over and over again.

Distortion Measurements depending on Frequency and Level

All measurement methods presented here can be stepped over frequency and/or level to obtain the THD data in relation to the frequency and level. This can either be displayed as a series of 2D plots or as one 3D plot.

While 2D plots are usually more suitable for displaying different kind of data (THD, THD+N, etc. for different channels) in one plot, the 3D display only works well for one specific selection and one channel, since the height information (e.g. THD value) also defines the colour for a higher readability. This makes it possible to read such a 3D diagram at the first glance on any 2D display and even to descry the height data in case of a top-down view.

For the traditional stepped pure-tone method, many individual measurements must be carried out depending on the desired spectral and dynamic resolution. This can take a long time considering the fact that a dead time between each measurement is necessary as well. For a frequency range of 20 Hz to 20 kHz with steps of 1/12 Octave and a dynamic resolution of 1 dB over a range of 50 dB a total number of 6000 single measurements have to be performed. In contrast to this it would only take 50 frequency sweeps or 120 level sweeps. Assuming a duration of each measurement of about 4 seconds, the traditional method will take 5 hours, while the frequency sweep will take only 2:30 min and the level sweep 6 minutes.

Furthermore the frequency sweep has a higher spectral resolution especially in the high frequencies, likewise the level sweep can have a higher dynamic resolution without additional costs in measurement duration.

Based on the measurement results the level sweep method delivers a more accurate representation of the THD, considering the fact that more harmonics can be taken into account and the use of window functions is not mandatory here. However, the frequency sweep has the advantage to deliver the transfer function along with the HIRs.

Conclusion

Within this paper the traditional method for measuring total harmonic distortion and two faster methods were presented. It was shown that a significant speed-up in measurement duration can be gained and that even the resolution in some aspects can be higher in contrast to the traditional method.

The proposed methods combined with a 3-D representation can be ideal for production line testing, where a quick decision is necessary.

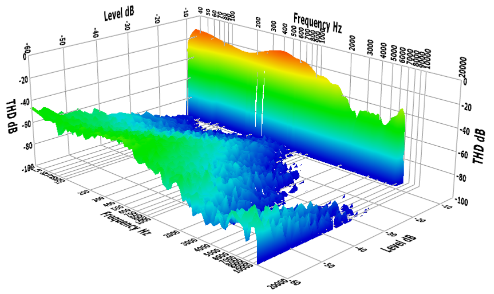

Figure 9: Traditional pure-tone stepped distortion measurement in a 3-D representation

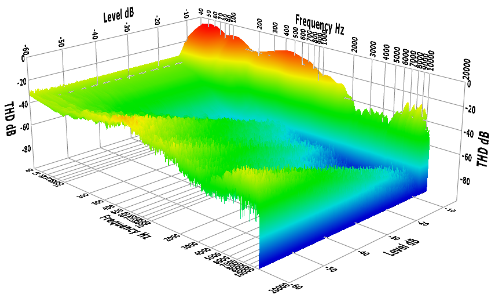

Figure 10: Frequency sweep distortion measurement in a 3-D representation

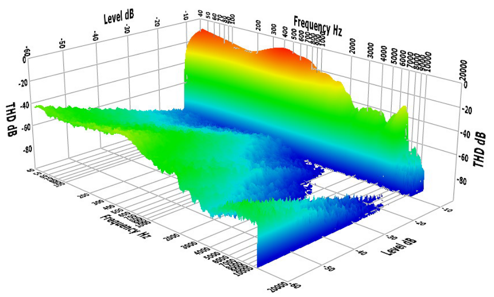

Figure 11: Frequency sweep distortion measurement in a 3-D representation

References

[1] Swen Müller, Paulo Massarani: Transfer-function Measurement with Sweeps, *J.AES*, 2001, p.443-471.

[2] Angelo Farina: Simultaneous Measurement of Impulse Response and Distortion with a Swept-sine technique, *J.AES*, vol. 48, p. 350, 108th, e *AES Convention*, Paris 2000, Preprint 5093.

[3] Swen Müller, Nelson Melo do Espírito Santo: Fast Distortion vs. Level Measurements with smoothly Increasing Signal Amplitudes, *5º Congresso de Engenharia de Áudio, 11ª Convenção Nacional da AES Brasil*, São Paulo, 2007