# Distortion Measurement at 10 MHz

Using Filters and a Spectrum Analyzer

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When it comes to high-precision signal measurement, you must stick with time-proven techniques. This article describes how to measure distortion at a frequency of 10 MHz. That's close to the upper limits of the audio range, right? But joking apart, if you can make a good job of handling 10 MHz signals, you can then also measure audio signals correctly. The latter is even easier, because spectrum analyzers for the audio region already have the right filters fitted. At 10 MHz you must roll up your sleeves and go hands-on.

Take the following scenario: you need a low-distortion op-amp for a project. You've identified a promising candidate and have even bought an evaluation board fitted with this IC from the supplier. Now you want to know what this amplifier does in reality and measure its distortion characteristics. According to the datasheet, the undesired

Ref 10 000 dBm RBW 10 000000 kHz VBW 10 000000 kHz Aften 15 0 00000 MHz - 51 050 dBm Mar 2 20 000000 MHz - 51 050 dBm Mar 2 20 000000 MHz - 51 050 dBm Mar 3 00 00000 MHz - 51 054 dBm Mar 5 50 000000 MHz - 71 244 dBm Mar 5 50 000000 MHz - 71 244 dBm Mar 5 50 000000 MHz - 55 564 dBm Mar 5 50 000000 MHz - 55 564 dBm Mar 5 50 000000 MHz - 55 564 dBm Mar 5 50 000000 MHz - 55 564 dBm Mar 5 50 000000 MHz - 55 564 dBm Mar 5 50 000000 MHz - 55 564 dBm Mar 5 50 000000 MHz - 55 564 dBm Mar 5 50 000000 MHz - 55 50 00000 MHz - 50 50 00000 MHz - 55 50 00000 MH

Figure 1: Spectrum of the 10-MHz signal from the signal generator, measured directly with the spectrum analyzer.

signal components produced by this op-amp at a signal frequency of 10 MHz and a signal level of 0 dBm lie in the region of -70 to -80 dBc (decibels relative to the carrier). Fair enough but what precisely does this mean?

# Clarification

First off, a level of 0 dBm corresponds to 223 mV $_{rms'}$  or with sinewave signals to 632 mV $_{pp'}$  into a load of 50  $\Omega$ . But that's not the only term we need to explain.

What does THD actually mean? We're referring to the acronym of Total Harmonic Distortion, which specifies the ratio of the sum of the powers of a number of harmonic signal components relative to the signal (carrier), that is to say simply a specific level ratio that indicates the total unwanted (disorderly) products in relation to the wanted signal. The THD is usually stated in the unit dBc, where the 'c' stands for carrier. How many harmonics should be considered is a matter of definition. In the MHz range it's the first and second harmonics that dominate principally; the third and fourth usually still figure well within the spectrum and the remainder frequently play no significant role. For this reason we confine our attention to the levels of the first four harmonics in practice. In the audio domain things can be different, on account of the increased quality demands for audiophile hardware (with 24-bit A-to-D conversion and 192 kHz sampling rate, etc.). Therefore, significantly more harmonics tend to be considered in audio electronics. Consequently a given THD value always includes an indication of

precisely how many harmonics have been considered. Of course, the THD value itself does not say anything about how individual power levels are distributed across each of the separate harmonics exactly. One particular harmonic may dominate, or else the energy may be spread more or less evenly across several harmonics.

## **Practical matters**

Let's get going then. For our example with the op-amp, we'll use a signal generator to produce a sinewave signal of 10 MHz with a level of 632 mV<sub>nn</sub> and connect its output to the input of the op-amp. To the latter's output we then connect a spectrum analyzer (SA). For our first examination of the resulting spectrum we set the analyzer's frequency range to cover from 5 MHz to 55 MHz. The screen of the spectrum analyzer might then look as seen in Figure 1.

Disillusion sets in already. The visible harmonics are clearly greater than what we'd wish to see in an op-amp of this kind. The spectrum above 55 MHz can safely be ignored though. A reason for the surprisingly high peaks in the harmonics could be quite simply that the generator itself is not great shakes and already includes a high harmonic content in its signal.

## Low-pass

Low-pass filtering of the output signal might be a fairly simple remedy. A corrective low-pass filter of this kind should allow the 10 MHz fundamental signal to pass with as little attenuation as possible, at the same time suppressing as effectively as possible the harmonics at 20, 30 and more MHz.

**Figure 2** shows the amplitude curve profile of a low-pass filter that was inserted in place of the op-amp. Before the signal from the sig-gen reaches the input of the spectrum analyzer, it needs to pass through a ready-built low-pass filter (type SLP-10.7 from Mini Circuits, see [1]). Of course, you can also construct a low-pass yourself, but a commercial one like this does after all offer guaranteed characteristics and at just over €30 / £25 / \$33 approx. is not outrageously expensive.

The markers are set at 10 MHz and the integer multiples of this figure. The attenuation levels at 30, 40 and 50 MHz lie in the region of -70 dB. If the sig-gen offers innately a signal quality of only 40 dBc, with this filter you would end up with a noticeably better 110 dBc at the second and successive harmonics, allowing you 'headroom' for measuring the expected 70 to 80 dBc of the op-amp sensibly. All the same, at 20 MHz the filter provides only 37 dB of attenuation. Here the sig-gen would need to come up with a signal-to-noise ratio of at least 53 dBc to retain some leeway of at least 10 dBc with the resulting 90 dBc. That is not enough.

What else could we do here? You could for instance simply raise the signal frequency (e.g. to 11.3 MHz), in the process also raising the attenuation of the harmonics in this filter. The frequency response of the filter in **Figure 3** indicates the effect of raising the generator frequency to 11.3 MHz. You'll see the attenuation of the first harmonic at 22.6 MHz is now 45.5 dB. Together with the 40 dBc signal-to-noise ratio of the sig-gen we hit the still almost usable 85.5 dBc of the test signal.

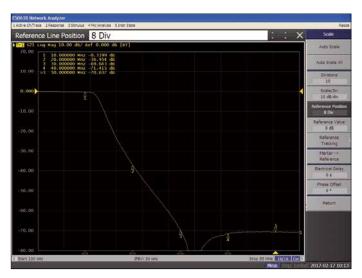


Figure 2: Amplitude curve of the SLP-10.7 low-pass filter by Mini Circuits.

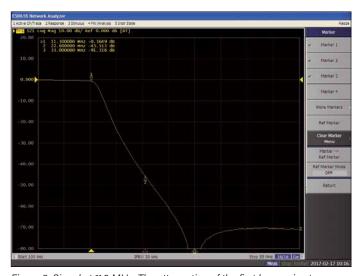


Figure 3: Signal at 11.3 MHz. The attenuation of the first harmonic at 22.6 MHz is now 45.5 dB.

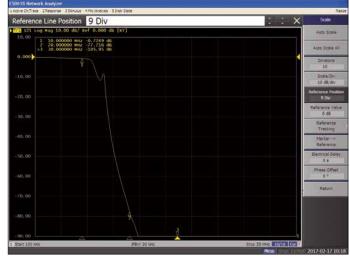


Figure 4: Frequency response of two SLP-10.7 filters in series.

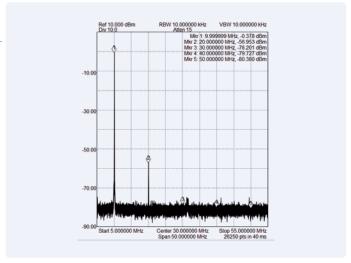


Figure 5: Spectrum after filtering by two SLP-10.7 low-pass filters.

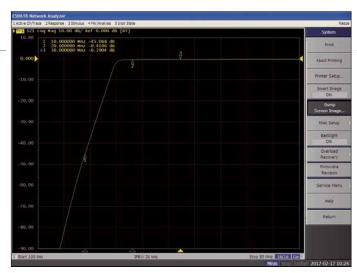


Figure 6: Frequency response of the SHP-20+ high-pass filter by Mini Circuits.

# **Doubling up**

For further improvement you could simply connect two low-pass filters of this type in series. Figure 4 shows how this would work out. Obviously the two filters demonstrate some mutual influence, making this arrangement not one to employ rationally for frequencies higher than 10 MHz. The attenuation of almost 78 dBc at 20 MHz has now increased to such an extent that sufficient scope remains for measurement.

It's vital to mention here that in the real world filters do not exhibit ideal properties but instead distortion and a tendency to create harmonics. The reasons for this lie in the less-than-ideal properties of the components in a filter. The coils in a 10 MHz filter usually have ferrite cores, which exhibit non-linear behavior. At elevated signal levels this can even lead to rising levels of distortion. Poor-quality capacitors can also introduce undesired signal components. These aspects are something you should definitely bear in mind if you consider building your own filters, otherwise you might end up making false economies.

It's interesting to examine the signal after passing though two SLP-10.7 low-pass filters in series (Figure 5). The harmonics at 30, 40 and 50 MHz are down in the noise, yet the peak at 20 MHz is now -57 dBm and has improved by only 6 dB. Why is that? It sounds amazing but in fact it makes sense: having less-than-perfect amplification in the input stage of the spectrum analyzer means that high signal levels can themselves produce harmonics as well. So we need a solution for this too.

# Add a high-pass

To mitigate distortion in the analyzer's input stage we can insert a suitable high-pass filter directly ahead of the input of the spectrum analyzer (i.e. between op-amp output and SA input). This high-pass reduces the level of the 10 MHz signal sharply. Figure 6 indicates the frequency response of a suitable high-pass filter (type SHP-20+ from Mini Circuits). The 10 MHz signal is attenuated by a full 45 dB, leaving the harmonics hardly affected. Figure 7 illustrates the changes made when the spectrum analyzer equipped with low-passes and the

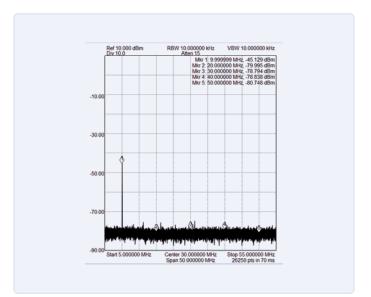


Figure 7: Signal spectrum after filtering by two low-pass and one high-pass filters.

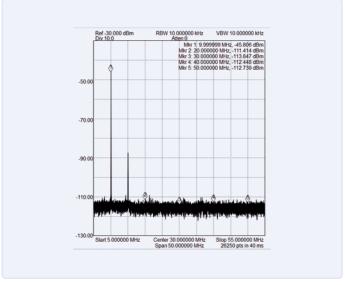


Figure 8: Spectrum with op-amp included and enhanced spectrum analyzer sensitivity.

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additional high-pass is driven by a quasi non-disruptive input signal. As expected, the level of the 10 MHz signal is now -45 dBm and the harmonics have disappeared in the noise. We can now adjust the analyzer's input stage to make it significantly more sensitive, since it is no longer possible to overdrive it with a too-high wanted signal now. The peak level in Figure 8 is now around -30 dBc. At the same time the noise floor of the spectrum is now reduced to below -110 dBc. With this setup we can now make quality measurements! The noteworthy peak at 15 MHz comes from the signal generator and is probably an artifact of the way it implements DDS (Direct Digital Synthesis). In order for us to measure the first harmonic at 20 MHz reliably, we can set the center frequency of the display to the harmonic (that is, 20 MHz) and reduce the span drastically (e.g. to 100 kHz). As a consequence the noise level falls to around -125 dBc, as the filter bandwidth of the spectrum analyzers is now much narrower. Figure 9 demonstrates strikingly that the first harmonic is now visible, although its level amounts to only -110 dBm. The noise margin here remains an acceptable 15 dB. The level of the higher-order harmonics is even smaller.

#### And finally...

Having now optimized our measurement arrangements so that we can actually detect the anticipated distortion in the op-amp, it is time to do the job properly. We do this in two steps: first we connect the output of the sig-gen, via the two low-pass filters, to the input of the op-amp. This needs to be terminated here with 50  $\Omega$ . The output of the op-amp then drives the input of the spectrum analyzer via a 50  $\Omega$  resistor

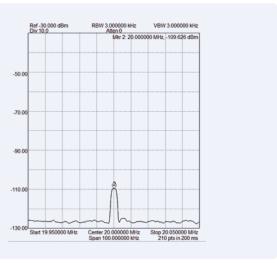


Figure 9: Zooming-in spectrally to 20 MHz ±50 kHz: the noise is reduced and the signal-to-noise ratio of the first harmonic at 20 MHz is now 15 dB.

(Setup 1, see Figure 10). The desired level of the 10 MHz test signal is adjusted on the sig-gen and measured using the spectrum analyzer, as this serves as a reference for 0 dB. Then we insert the high-pass filter between the op-amp and the spectrum analyzer (Setup 2) and quantify the harmonics. Everything is good now.

## **Last words**

In addition to the three filters, a Siglent SDG1025 signal generator and Signalhound SA44B USB Spectrum Analyzer were used as the measuring instruments.

(160427)

# WEB LINKS .

- [1] www.minicircuits.com/WebStore/dashboard. html?model=SLP-10.7%2B
- www.minicircuits.com/WebStore/dashboard. html?model=SHP-20%2B

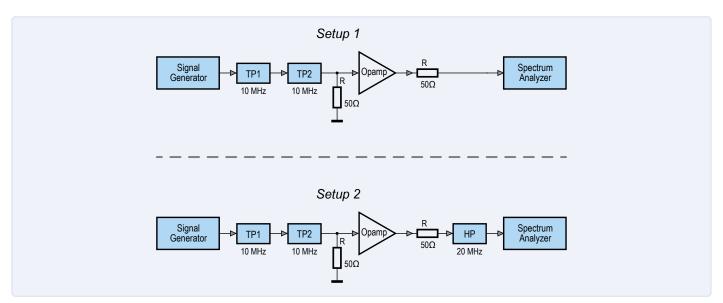


Figure 10: Two-step measurement procedure. Setup 1 measures the reference level and Setup 2 the harmonics.