

Multi-Tone Signal Generation with AWGs

White Paper

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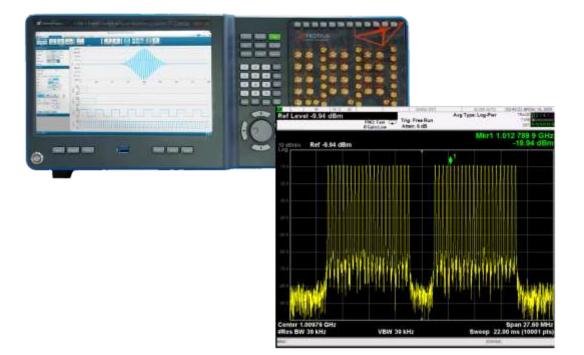




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1 Generating Multi-tone Signal with AWGs

AWGs can generate a fully defined baseband (I/Q complex signal with two channels) or an IF/RF (real signal with just one channel) multi-tone signal from a mathematically defined waveform. Continuous generation is possible thanks to the seamless looping capability existing in all arbitrary waveform generators. No matter the size of the waveform memory, eventually, the waveform will have to be looped in order to keep the signal alive as long as any test may require. Calculating the samples to be stored in the memory is as simple as applying the following expression (see <u>Error! Reference source not found., page Error! Bookmark not defined.</u> for a MATLAB script using the multi-tone calculation methodology shown below):

$W(n) = S A(i) * \cos(2 * p * (f0 + i * Df) * n / SR + f(i)), i = 0,...M-1, n = 0,...N-1$ (1)

However, successful generation of a multitone signal requires meeting some conditions to make sure that it can be looped seamlessly. In order to obtain a continuous, seamless looping, it is enough to make sure that for all the tones, an integer number of cycles will be stored in the available waveform memory. This can be accomplished by choosing the right sample rate, waveform length and, if necessary, modifying the tones' frequencies while keeping equal spacing for all of them. The best way to define the signal generation parameters is, probably, to define the target Sample Rate. The basic Period (P) of the multi-tone signal can be calculated from the Δ f parameter:

P = 1 / Df

The initial Time Window (TW) to be implemented in the waveform memory can be any integer multiple of the Period P:

TW = k * P, k = 1, 2, 3...(3)

The number of samples to implement the TW (RL₀) depends on the Sample Rate (SR):

WL0 = round(TW * SR) (4)

The rounding to an integer in the above expression results in the need to modify the Df parameter so an integer number of periods are stored in the actual waveform length. The actual time window (TW') will be:

TW' = WL0 * SR (5)

And the actual spacing (Df') will be:

Df' = Df * TW/TW' (6)

All high-speed AWGs require the actual record length to be a multiple of some integer (G or Granularity). So, the actual waveform length (WL_1) will have to be modified to meet the granularity requirement. A simple way to fulfill this requirement is by rounding the waveform length to the nearest valid integer:

WL1 = round(WL0 / G) * G(7)

The above expression results in a coarse rounding of the waveform length so the actual Δf after applying (5) and (6) will be less accurate. A different way to solve the granularity problem is by repeating the same basic waveform length as many times as required to obtain a multiple of G so the $\Delta f'$ calculated in (5) will be kept. This can be calculated through the minimum common multiple (mcm) operator:



WL1' = mcm(WL0, G)(8)

So far, just the record length and the actual Δf have been obtained. The other parameter that must be adjusted is f_0 so an integer multiple of cycles are stored for all the tones. One of the ways to fulfill this requirement is choosing a reference tone (typically the first one, the last one, or the one in the middle) and obtaining the closest frequency meeting the criteria. The reference frequency must be calculated with the original parameters:

$$Fn = f0 + n * Df$$
 (9)

Any periodic signal with an integer number of cycles for a given time window must have a frequency that must be a multiple of:

Fr = SR / WL1(10)

The reference frequency must be rounded to the nearest multiple of that frequency:

Fn' = round(Fn / Fr) * Fr (11)

Once the reference frequency has been adjusted and using the tone spacing already calculated, the frequency of the first tone can be obtained:

F0' = Fn' - n*Df' (12)

Final frequencies for all the tones can be obtained:

Fi '= F0 ' + i * Df', i = 0...N-1 (13)

As the amplitude and phase for each tone does not affect the looping continuity behavior of the resulting signal, the samples for the waveform can be calculated using expression (1) with the frequency values listed in (13). As a rule of thumb, the longer the waveform length, the lower the frequency errors for the tones. Selecting a high k in expression (3) and choosing the granularity handling strategy shown in expression (8) will be better to reduce frequency error. If the resulting frequency errors are not acceptable, $\Delta f'$ can be corrected by modifying the sampling rate (SR'):

SR' = SR * TW' / TW, so Df' = Df (14)

In this case, f0 must be recalculated as well using expression (11).

Real-world AWGs will not generate ideal multi-tone signals, even at those corrected frequencies. They will be affected by linear and non-linear distortions and different noise components. Linear distortions are a consequence of the non-flat amplitude, nonlinear-phase frequency response of any arbitrary waveform generator.



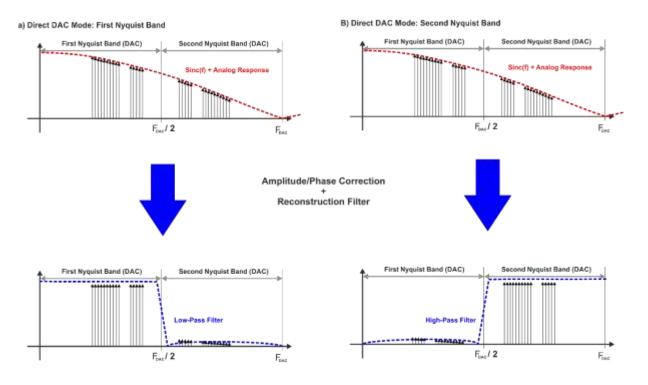


Figure 1.1 The linear distortions added by the AWG frequency response can be characterized and corrected in the first Nyquist Band (a) or the second (or higher) Nyquist band (b). This correction results in a high-quality multi-tone signal with a very accurate statistical behavior (PAPR, CCDF). It is not possible to correct the multi-tone signal simultaneously in the first and second Nyquist band without using external filters. For this reason, multi-tone signals must always fit in one of the Nyquist bands.

Overall frequency response is made of two components. One is the combined frequency responses of the DAC, the output stage, and any cabling, component and fixturing attached to the output. This component does not depend on the sampling rate. The second component is the zeroth-order hold response of the DAC. Theoretically, this frequency response for amplitude takes the shape of a sinc(k*f) with all its zeros located at the sampling frequency and its multiples while phase response is linear as it only adds some constant delay. Multi-tone signals are an excellent method to measure the overall amplitude/phase frequency response if a flat response analysis instrument is available, so correction factors can be obtained through a calibration procedure. Spectrum Analyzers (SAs) usually offer a good frequency range, a good flatness, and an excellent dynamic range. However, these instruments are not sensitive to phase, so phase response cannot be established. Maximally flat response DSOs (Digital Storage Oscilloscopes) are also a good alternative. These scopes have corrected amplitude and phase response themselves, so they show an excellent flatness and phase linearity, and frequency-domain measurements are possible through the usage of the built-in FFT function. However, they do not offer an extremely good dynamic range. Spectrum analyzers are the best choice to analyze multi-tone signals when only amplitude correction is required. However, DSOs should be the tool of choice if both amplitude and phase must be corrected. Correction factors can be used to correct the results at the



output of the DUT, or they can be used to correct the multi-tone signal being generated, so the output will reflect the actual frequency response of the DUT. Expression (1) should be extended to include these corrections:

$W(n) = S C(i) * A(i) * \cos(2 * p * (f0 + i * Df) * n / SR + f(i) + fc(i))$ (15)

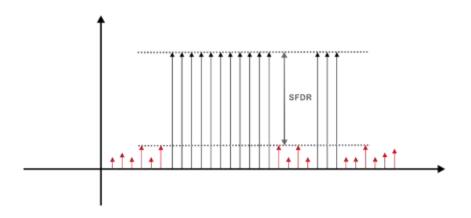
It is important to understand that linear corrections will typically boost the higher frequency components, and it will modify the PAPR of the corrected signal. The most likely effect will be that the flatness will be better, but the overall signal power will be reduced and, therefore, dynamic range will also be reduced. Correcting multi-tone signals where differences between tones are larger than 10/15 dB may result in low-quality signals.

Non-linear behavior has also two main components. One of them is the non-linear nature of the DAC itself, even when it is ideal as quantization is a non-linear operation. In addition to quantization, the effects of INL (Integral Non-Linearity) and DNL (Differential Non-Linearity) will also add to the effects. The second component is the non-linear response of the output stage and any active component attached to the output of the generator. In order to use multi-tone signals as test signals, to check the non-linear behavior of some DUT, the non-linearities of the signal itself must be much lower than those expected to be measured in the DUT. External components cannot be controlled by the AWG (although they may be corrected as it will be seen in the next section), so the internal processing blocks and components, are going to be analyzed here. The output stage is especially important as today's state of the art DACs can show an excellent linearity. Some AWGs, such as the Proteus series, can alternatively incorporate an output amplifier or a direct, unamplified connection from the DAC. Difference, in terms of SFDR and IMD products, may be huge in favor of the second choice.

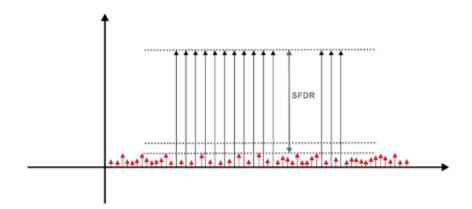
Quantization noise is one of the unwanted signal components, see Figure <u>1.2</u> a). When generating a periodic signal, like a multi-tone signal in a seamless loop, quantization noise also repeats for each loop so its spectrum will be made of a series of tones located at multiples of the repetition frequency. When only one period of the multi-tone signal is stored in the waveform memory, quantization noise will show up exactly where tones and regular IMD products are located, including notches. The solution to this problem is setting k in expression (3) to a large number, and making sure that the different periods in the waveform memory never repeat exactly, see Figure <u>1.2</u> b). One way to make sure this condition is met is assigning a high-enough primer number while making sure the waveform length is not a multiple of it. Quantization noise tones will be still there but as there will be many more of them and the overall power stays the same, the average power for them will be lower. IMD caused by amplifiers and INL will keep the same period of a single multi-tone period, they will show up as regular IMD products, though. Another way to greatly reduce the effects of quantization noise is by using the AWG's internal DUC. As the carriers generated by the internal NCOs (Numerical Controlled Oscillators) are not synchronous with the I/Q waveforms, the output signal does not repeat exactly in the same way at each loop of the waveform memory and this "spreads" out quantization noise, see Figure <u>1.2</u> c).



a) Multi-Tone Generation: One repetition period (k = 1)



b) Multi-Tone Generation: Two repetition periods (k = 2)



c) Multi-Tone Generation: Internal DUC

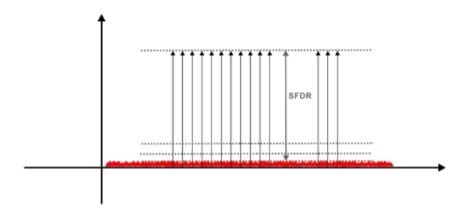




Figure 1.2 Quantization Noise is a component of any signal generated by an AWG. In AWGs, continuous signals, such as multi-tone signals are generated by looping a self-consistent fragment stored in the waveform memory. As this segment repeats, quantization noise will also repeat, and it will show up as additional tones in the signal's spectrum a). If the same section is calculated in such a way that several non-exactly equal segments quantization noise will repeat over a long period and more tones with a lower amplitude will show up, so the SFDR will improve b). Using an internal DUC is equivalent to applying real-time dithering and the discrete tones will disappear from the spectrum, and replace by a dense, continuous background noise, so SFDR will improve even further b).

The above distortions and noise sources are part of the ENoB (Effective Number of Bits) specification. It tries to express the quality of some AWG (or a DAC) as that of an ideal DAC of some resolution (<u>Error!</u> <u>Reference source not found. Error! Reference source not found.</u>, page <u>Error! Bookmark not defined.</u>). However, the impact of the different components affecting ENoB may be very different on a multi-tone signal so this parameter by itself it is not a good enough differentiator between products.

Signal quality in terms of IMD products and SFDR can be optimized following these guidelines:

- 1. Select the highest possible sampling rate as this improves flatness and spreads quantization noise over a larger BW so its noise power density is reduced.
- 2. If possible, make sure that the sampling rate is selected in such a way that the image of any harmonics of the fundamental multi-tone signal is located far from the band of interest.
- 3. Select a large prime number of multi-tone cycles and make sure the waveform length is not a multiple of it.
- 4. Use the best possible phase distribution to obtain a good enough PAPR so the maximum overall power is obtained while keeping the target peak power. Avoid constant distributions.
- 5. Use the best possible output stage if no linearity corrections are available.
- 6. Make sure external components such as amplifiers, mixers of modulators are up to the task.
- 7. Use a linear correction tool to compensate for the frequency response of the AWG and any components or block attached to it.
- 8. Apply non-linear corrections if possible (see <u>Error! Reference source not found. Error!</u> <u>Reference source not found., page Error! Bookmark not defined.</u>).



Resources & Contact

For more information on microwave signal generation challenges and solutions, review the following resources:

- White Paper: <u>Multi-Nyquist Zones Operation-Solution Note</u>
- Solution Brief: <u>Envelope Tracking Solution Note</u>
- Datasheet Proteus Arbitrary Waveform Generator / Transceiver: Module Desktop Benchtop
- Datasheet Proteus RF Arbitrary Waveform Generator / Transceiver: Module Desktop Benchtop
- Online Webinar: Experiment design considerations for real-time, closed-loop pulse streaming

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