

Time Matters — How Power Meters Measure Fast Signals

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Power Measurements

Modern wireless and cable transmission technologies, as well as radar systems, present demanding challenges for device and system developers. Manufacturers of test and measurement equipment are driven to offer products that fully support today's needs, while anticipating the requirements of future technologies. Accuracy has always been a critical requirement in the test and measurement world, but modern technologies demand another must-have — highest data acquisition and processing speeds to allow accurate measurements of complex signal waveforms. This paper describes the different techniques RF peak power meters employ to meet these challenges.

Signal Triggering

Modern peak power meters can measure virtually all types of pulsed or repeating signals. To achieve this, these instruments are equipped with sophisticated trigger capabilities. Prerequisite to any fast measurement is the synchronization of the instrument's measurement cycle to the actual event. Simply put, the input signal of interest has first to be "found." Specific trigger settings prepare the instrument for this synchronization and, once the desired event occurs, provide stable signal representations, allowing detailed signal analysis and accurate measurements. To be able to "look ahead," digital instruments often use special techniques such as circular acquisition buffers to facilitate display and measurement of pre-trigger events. Most RF peak power meters provide internal and external trigger capabilities. Internal triggering utilizes the envelope of the actual incoming RF signal, while external triggering utilizes a baseband trigger signal that is in some way synchronized with the RF input signal.

Data Acquisition

Data acquisition systems for fast analog signals typically consist of sample and hold (S&H) circuits, analog-to-digital converters (ADCs), digital signal processors (DSP) or field-programmable gate arrays (FPGAs), and processing and interfacing units. Each of these function blocks needs a finite period to convert or process data, which is known as latency. While smart methods, like staggered buffering, can temporarily reduce data acquisition time, the sum of the latencies basically define the performance of a power meter in regard to maximum sampling frequency, also commonly known as sampling rate or sustained sampling rate.

Analog-to-digital conversion of fast continuous signals results in a stream of discrete data points. These data points contain only limited information: sampling point of time, determined through the position in the memory, and its value. For power measurements, each data point represents only a "slice" of the original signal, but slices are not sufficient for in-depth analysis and accurate measurements. Let's examine various methods how this limitation can be overcome.

Conventional Interpolation Methods

Interpolation is a "connect-the-dots" processing technique used to estimate what a waveform could look like, based on a limited number of sample points. The simplest interpolation method is linear. It is difficult, however, to obtain acceptable reading using linear interpolation, unless the sampling rate is very high. Obviously, this puts a higher burden on the data acquisition and processing circuitry, hence raising the equipment costs. Are there alternative ways to connect the dots?

DSPs allow applying fast Fourier transformation (FFT) techniques to rebuild sampled waveforms. Joseph Fourier spearheaded the mathematical proof that any waveform can be reconstructed from sinus functions equal to its basis frequency and its multiples (harmonics). Hence, using interpolation of discrete data points through a $\sin(x)/x$ function provides a much more refined rebuild waveform with closer resemblance to the original signal, as shown in *Figure 1*.

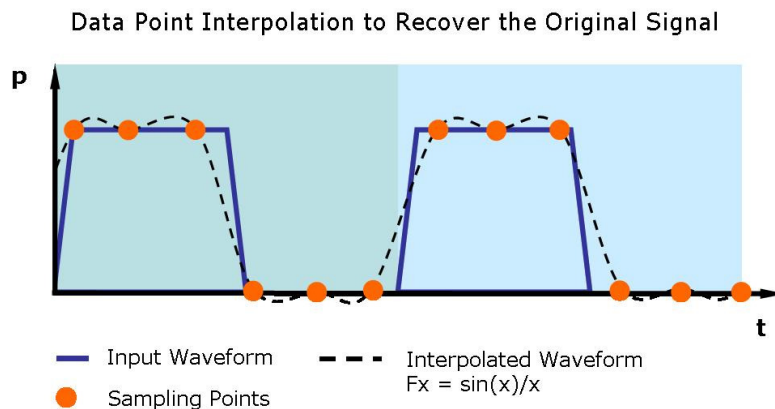


Figure 1: Interpolation of a waveform via $\sin(x)/x$ function. The sampling rate is 3 times the Nyquist frequency f the input signals basis frequency, which comes however with a high harmonic content.

What sample frequency is required to reconstruct the original signal? Harry Nyquist and his colleagues propose an answer: Nyquist's sampling theorem states that the minimum sampling frequency of a limited bandwidth, time-continuous signal may be no less than twice the maximum signal frequency in order to fully reconstruct the signal from the acquired discrete data.

Nyquist's theorem also applies for non-baseband signal frequencies with a limited bandwidth. The required sample frequency then depends on the signal's bandwidth. As with baseband frequencies, the sampling rate must then be higher than twice the occupied bandwidth. As an example, a signal with a bandwidth of 5 MHz would require a sampling frequency of just above 10 MHz to provide a sufficient

number of data points to fully reconstruct the signal — at least in theory. We need to consider that the theorem is a mathematical model and based on ideal conditions. Furthermore, Nyquist requires all data points following the relevant one to be taken in consideration, which is also rather impracticable. Realistic approaches could use data points 1 through 8 for the first interpolation, 2 through 9 for the second, and so on. Real-world signals have another typical characteristic: Their amplitude may vary dramatically within a very short time. Fast variations generate additional harmonics (Fourier!). A prominent example would be a rectangular signal that consists an infinite amount of sine wave multiples of the base frequency. Since Nyquist's theorem does not allow higher frequencies than half the sampling rate, these harmonics have to be completely suppressed. While today's filter technology is quite advanced, it is simply not practical in realizing filters with brick-wall characteristics, filters that would completely suppress all signal energy below and above the bandwidth of the measured signal while allowing 100% of the actual signal to pass.

Suppressing harmonics before the sampling process does alter the waveform that will be sampled (Fourier again!). Depending on the original waveform, this can be insignificant; but if too much energy is cut off, the reconstructed signal will show only degraded resemblance to the original signal, thereby reducing measurement accuracy. Since filtering out upper signal harmonics removes a portion of the signal's total power, not only will the reconstructed waveform appear incorrect, but the total RF power measured by the instrument also will be low.

If signal elements higher than Nyquist frequencies are permitted to pass through the input filter, and are sampled by the ADC, resulting data points are undistinguishable between wanted and unwanted content. The total RF power will now be correct, but reconstruction of the sampled waveform will cause alias effects, creating jitter and reducing displayed accuracy significantly. The sampling rate in power meters needs to be much higher than double the Nyquist frequency since highest-order filters are very difficult to realize, especially with low and stable insertion loss. Furthermore, they tend to roll off important portions of the waveform. A more common approach is to use higher sampling rates along with lower-order filters.

Repetitive Random Sampling

Repetitive random sampling (RRS) is not to be confused with averaging. RRS, as shown in *Figure 2*, is a technique in which power meters construct a full picture of a repetitive signal by capturing little bits of information over several repetitions of the event. This is accomplished by using an internal clock that runs asynchronously to the actual signal trigger. The power meter takes continuous samples independent of the trigger event. Although the samples are taken sequentially in time, they are always completely random with respect to the trigger. Data points are added with every sweep. Depending on display time resolution set at the power meter, just one or a set of evenly spaced sampling points are added per sweep. As a result, the waveform is completely reconstructed. The screenshots in *Figure 3* show a set of three additional sampling points per sweep.

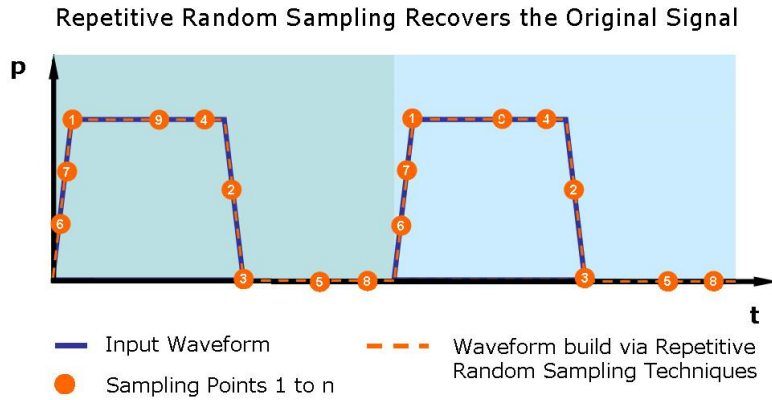


Figure 2: Repetitive random sampling (RRS) with small time increments rebuilds a waveform very close to the original waveform.

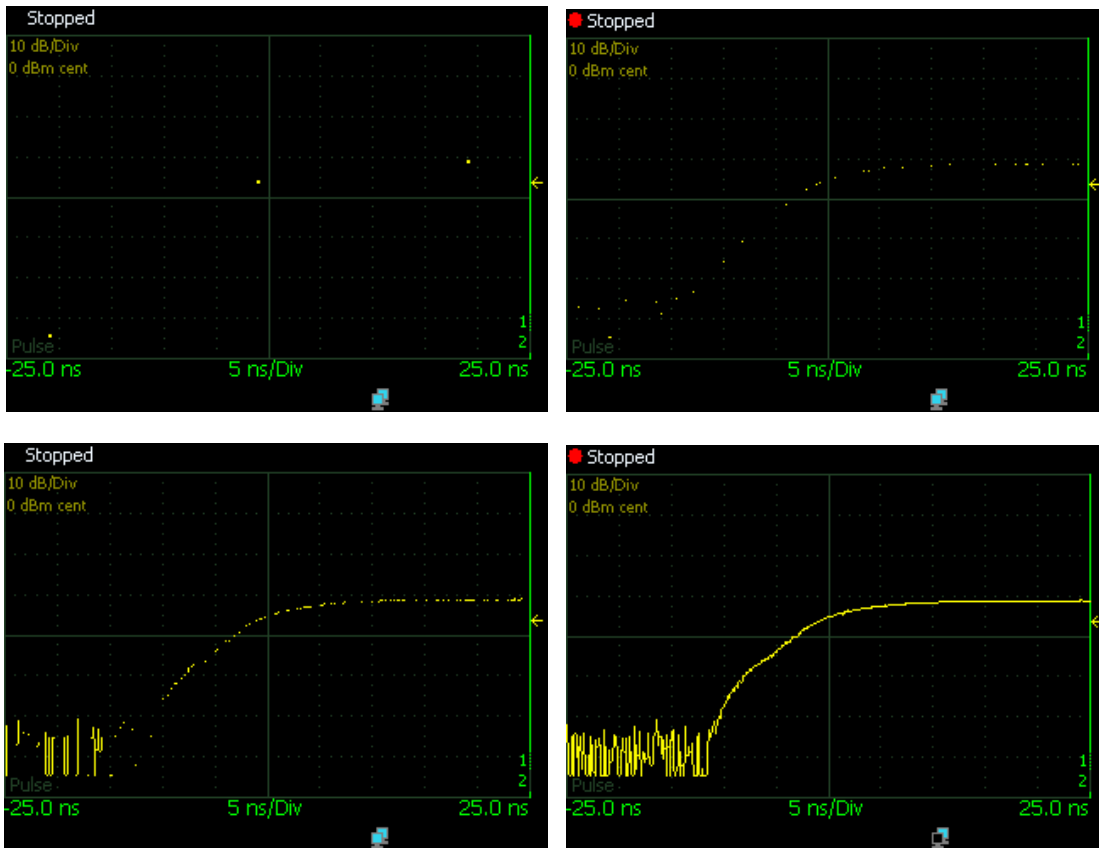


Figure 3: These four screenshots show how a waveform is build through repetitive random sampling techniques. The first sweep shows an initial set of three data points equally 20 ns apart. The remaining three show 10, 50, and 200 sets of additional data. This method achieves highest resolutions, allowing “zoom in” to fast signals.

Repetitive Random Sampling is limited by the smallest time increment that the instrument is capable of resolving. This is not to be confused with the sampling rate, which may or may not depend on the

instrument’s time increments. The general rule is: the smaller the increments, the more accurate the representation of the waveform will be. The instrument’s sweep time determines how long it takes to build the full waveform. With modern peak power meters, the time to achieve full representation is often dictated by the repetition frequency of the actual signal and not so much by the instrument’s performance. Compilation of the full waveform is achieved within milliseconds for typical input signal repetition rates.

The effective sampling rate that is achieved by RRS often reduces or eliminates the need for sharp anti-aliasing filter, along with the measurement inaccuracies imposed by this filter. For example, if an effective sample rate is 5 GSamples/s, then a first-order filter at 200 MHz will be more than sufficient to reduce sampling artifacts, while having a minimal measurable effect on signals even with harmonics 100 MHz and above.

RRS can increase the waveform display resolution by magnitudes. In *Table 1*, we have a practical comparison of how time increments influence the maximal resolution. Let us look at two peak power meters, which we will call PM1 and PM2, measuring a repetitive 50ns pulse. PM1 offers 10 ns time increments, and PM2 operates with 200 ps time increments. *Table 1* illustrates that PM1 would acquire 5 samples of the pulse, while PM2 would be able to acquire 250 samples. PM2 would provide 50 times higher resolution, allowing a more detailed analysis. Simply put, the latter instrument could “zoom in” 50 times compared to the former. Such high resolution is particularly relevant when ramp-up behavior, burst pre-shoots, burst over-shoots, filter characteristics, or output behavior of high-gain RF amplifiers need to be analyzed.

Power Meter Time Increments	Repetitive Random Sampling Rate (GSamples/s)	Acquired Data Points (repeating 50ns Signal Pulse)
10 ns	0.1	5
1 ns	1	50
200 ps	5	250
100 ps	10	500

Table 1: Comparison of displayed signal resolution depending on power meter time increments.

Conclusion

Power meters with higher system sampling rates provide an advantage in single shot/single sweep applications. The rule is that doubling the sampling rates translates in double the resolution or improved measured/displayed signal to noise ratio up to 3 dB under ideal conditions. Higher RRS rates require minimally longer building the display but offering a significantly higher waveform resolution and better, more accurate analysis of fast repetitive signals. RRS provides the user with details that may be missed when using conventional sampling methods.