Probability of intercept - what signal can I see?

By Paul Holes, Field Application Engineer, RFuWave Group

When examining signal behaviour using test equipment, there is some information we need to know:

- When and where does the signal occur, and for how long?
- What are its centre frequency and bandwidth?
- Is there any modulation and, if so, in what form?

It is possible to search in either frequency and/or time domain; however, the following should be borne in mind when deciding how best to analyse the signal of interest:

The frequency domain is best for identifying signals in the frequency spectrum: for example, is the signal dynamic or static, is it bursty or modulated, or is it just a continuous wave.

The time domain is best for identifying the occurrence of events, perhaps around a specific time or in relation to other events.

In the frequency domain, a spectrum analyser must possess the following criteria to be able to see the signal being analysed or hunted:

- It must have a higher frequency with an adequate dynamic range and a bandwidth wider than the signal of interest.
- Ideally, it should be capable of vector signal analysis and have a signal capture memory to repeatedly reproduce the signal, as well as to enable analysis of other device behaviour when subjected to the signal of interest.

Probability of intercept

Once we have established the type of signal to analyse or look for, what's the likelihood we will see it and for how long? This is called "probability of intercept", or POI. POI is a key specification to detect bursty or transient signals; it's defined as the minimum duration of a signal that can be observed with 100% certainty and without amplitude errors. Factors that determine POI are:

1) Sampling rate;
2) Time record length (or FFT size);
3) Windowing function and size;
4) Overlap processing;
5) Noise floor:

$$T_{Sweep} = \frac{SPAN \cdot k}{RBW^2}$$

where RBW is resolution bandwidth, T_{Sweep} is sweep time and k is filter skirt factor (2-3 for Gaussian filters).

$$POI = \frac{(R + T)}{(R + R')}$$

Here, T is duration of the signal of interest, R the listening time at frequency, R' the time not listening and R+R' is the revisit time.

$$\sim R = \frac{(RBW + S_{BW}) \cdot T_{Sweep}}{SPAN}$$

Also, S_{BW} is the spectral width of the signal.

Swept spectrum analyser principle

So, what happens when using a swept spectrum analyser? First, we need to understand the basic block diagram of the superheterodyne receiver in this kind of analyser.

The input signal is converted to an intermediate frequency (IF) by a local oscillator (LO) and a mixer. This signal is then swept past a fix-tuned filter to determine the resolution bandwidth, before being logarithmically amplified and passed on to the display.

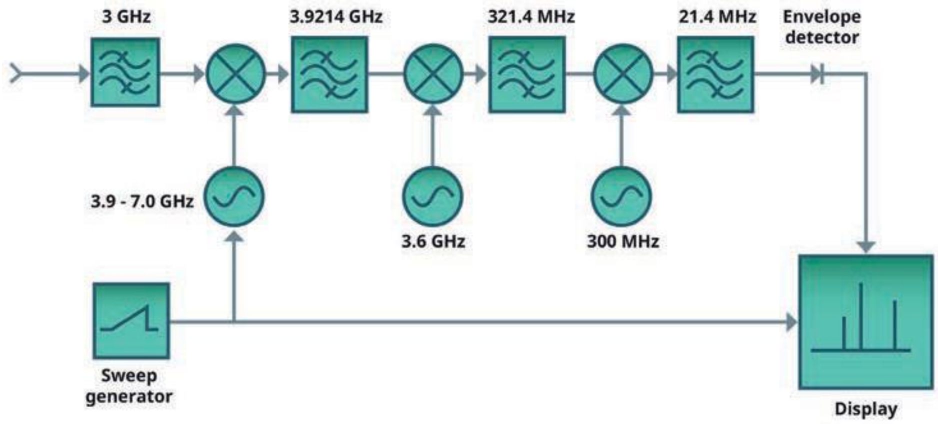
In Figure 2, the spectrum analyser is sweeping across the display, and if the green line crosses the signal, it means the signal can be seen on the display. However, the spectrum analyser also misses signals during its sweep retrace time. This is called "blind time".

POI can be improved by increasing the sweep speed, resulting in a higher number of signal crossings, but there is always a retrace time and therefore blind time.

The FFT spectrum analyser principle

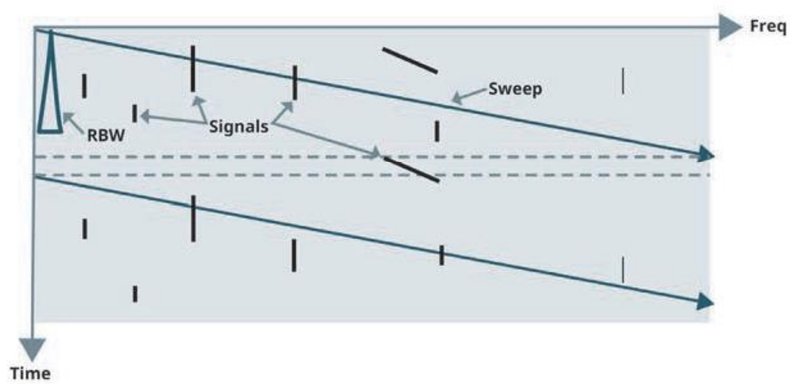
The FFT spectrum analyser is similar to the swept spectrum analyser; it has an analogue front-end but uses digital IF and processing. With the swept spectrum technique, the RBW filters were limited in their flexibility and therefore place fixed conditions for a swept spectrum analyser to resolve the required signal.

Figure 1: Basic superheterodyne receiver



$$f_{Sig} = f_{LO1} - (f_{LO2} + f_{LO3} + f_{IF})$$

Figure 2: Probability of intercept for swept spectrum analysis



An FFT digital analyser, which has decimating filters and resampling algorithms, provides a solution to the limited frequency resolution problem. Digital decimating filters and resampling perform the operations necessary to allow variable spans and resolution bandwidths. These filters simultaneously decrease the sample rate and limit the bandwidth of the signal, hence providing alias protection.

The sample rate into the digital filter is f_s and out of the filter f_s/n, where "n" is the decimation factor and an integer value.

Frequency spans that result from "divide by 2n" are called "cardinal spans". Measurements performed at cardinal spans are typically faster than measurements performed at arbitrary spans due to reduced DSP operations.

Since the waveform is analysed digitally, it can be captured in a relatively short time and then analysed. This short capture time can have many advantages, particularly with transients or short-lived waveforms. It also means the FFT analyser can capture non-repetitive waveforms, which is not possible with other spectrum analysers. As part of the signal capture process, data is processed to reveal the phase of the signals. This FFT technology also allows the waveform to be captured for later analysis, should it be required.

However, there are disadvantages and limitations associated with FFT technology. The main limitation on the frequency and bandwidth of FFT spectrum analysers is the A/D converter (ADC) that changes analogue signals into digital format. This component still has a limited upper frequency (and BW), even if down conversion has been applied and, because of its high performance, it's costly, especially since it also requires associated processing and display circuitry.

Stepped FFT principle

To achieve wider bandwidths than the IF, multiple FFTs are used together. With a high-speed local oscillator (LO) and a digitiser list mode, fast stepping is achieved across the whole span. The software stitches together the FFTs and displays a single trace result up to several tens of GHz.

First, the signal input to a spectrum analyser is converted to an intermediate frequency by the frequency conversion section that combines a local oscillator, mixer and bandpass filter. In a signal analyser, the LO's frequency is fixed during measurement.

Next, the IF-converted measured signal is converted to digital data using an ADC; i.e., the signal analyser captures the

Figure 3: FFT spectrum analyser analogue front end with digital IF processing

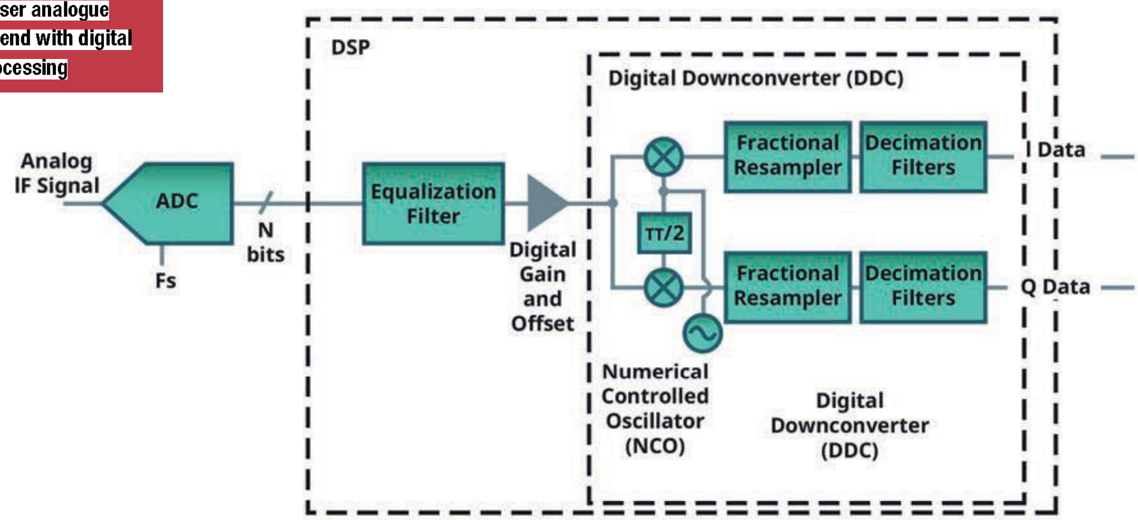
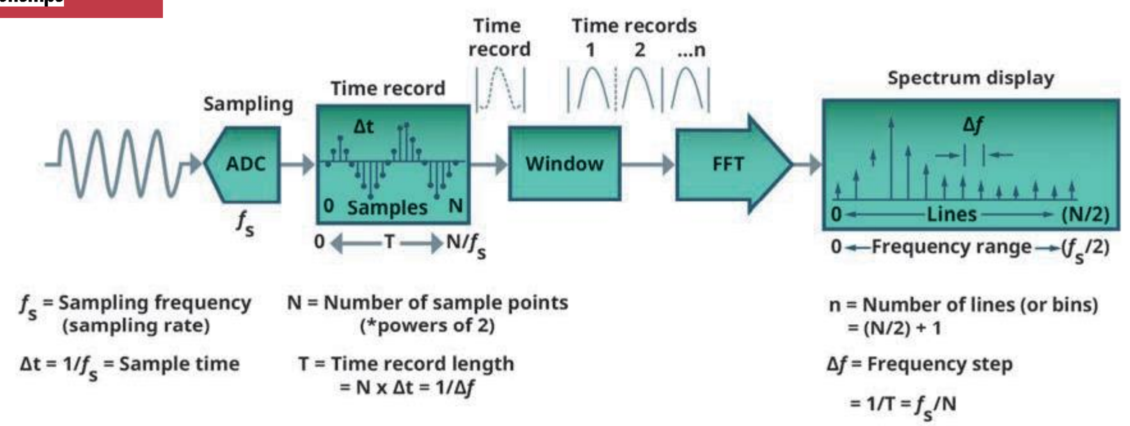


Figure 4: Basic FFT relationships



true form of the measured signal in a fixed time period just by frequency conversion. The digitised time series waveform data is immediately captured to internal memory (this data can also be saved to another hard disk). Any part or range of the measured signal captured as time-series waveform data can be read immediately and analysed using digital processing. Deploying digital processing using FFT in the frequency domain captures the spectrum in the read time range.

Along with conventional analogue modulation schemes such as AM, FM and PM, digital modulation can also modify the same, either separately or simultaneously. More commonly, however, vector modulation or I-Q modulation is used instead. Any arbitrary carrier phase and magnitude can be generated via vector modulation. The baseband digital information is separated into two independent components: the I (in-phase) and Q (quadrature) components. These I and Q components are then combined to form the baseband modulating signal.

Only phase and magnitude are required because the carrier frequency is fixed in modern (digital) communications systems. This is the phase and frequency reference. The modulated signal is interpreted relative to the carrier. The phase and magnitude can be represented in polar or vector coordinates as a discrete point in the I-Q plane, where I represents the in-phase (phase reference) component and Q represents the quadrature (90° out of phase) component.

Encoded information can then be transmitted by forcing the carrier to one of several predetermined positions in the I-Q plane. Each position or state represents a certain bit pattern that can be decoded at the receiver. The mapping of the states or symbols when the receiver interprets the signal on the I-Q plane is referred to as a constellation diagram. This is the principle of I-Q modulation.

The data stored in memory or hard disk is the basis of the signal analyser display. In other words, the signal capture processing that stores the signal as digital data, and the analysis processing that reads the data can be executed independently, time-wise. Since capture and analysis are batch-processed, either offline analysis can be performed in the free time after capture, or the same signal can be analysed repeatedly using different methods and settings. This is a key feature of signal analysers that is not shared by sweep-type spectrum analysers.

The RTSA principle

Real-time spectrum analysis (RTSA) is a method that uses overlapping FFTs and high-speed memory to achieve a 100% POI.

The phrase "real-time analysis" means different things to different people, but, essentially, in a spectrum or signal analyser with a digital IF, it is a state in which all signal samples are processed continuously and gap-free to derive a measurement result.

T refers to the analyser's time record for FFT processing ("n" FFT points); the analyser collects a block of contiguous samples from the ADC that digitises the IF signal and then performs an FFT to obtain a power or vector spectrum. It is important to understand that the T-block's time length will be shorter when bandwidths are bigger and sampling faster.

A loss of data refers to the frequent case where the samples in a time record are "windowed", which means attenuated at the beginning and end of the record, to avoid spectral leakage or to form the equivalent of different resolution bandwidth filters. This attenuation of data at the edges of the time record is often so significant that some samples are effectively lost. The solution to this problem is called "overlap processing".

Overlap processing provides significant benefits for signal analysis when processing is faster than sampling, by performing additional FFTs with partially-new time records as samples come in.

Figure 5 shows a 50% overlap where each new FFT is performed on 50% new samples and 50% from most recent samples of the previously-processed data. With significant overlap, it is possible to configure analysis so that all samples are processed for some FFT result with no or minimal attenuation or de-emphasis due to windowing. Thus, overlap improves POI and accuracy.

In summary, an RTSA has a fixed LO with a given IF BW, collecting IQ data over a certain period. The data is corrected and FFT-processed in parallel. Vector information is lost, and the units have very advanced displays for large amounts of FFTs.

Pros and cons

To summarise, each type spectrum/signal/RTSA has advantages and disadvantages. Compared with modern signal analysers, the traditional swept spectrum analyser has a superior frequency and an associated core-specifications set, but struggles to analyse transient and changing signals, and does not measure phase.

The signal analyser measures phase and pulsed signals, and can sample and replay waveforms quickly. It has an extensive set of demodulation capabilities, including I.

The RTSA measures multiple simultaneous waveforms, and displays signals and events more effectively by capturing the signal and using overlapping. However, because of ADC limitations, it generally has a smaller BW and poorer dynamic range.

The Anritsu MS2840A spectrum/signal analyser is a good example of using these features in their most optimum range to determine the characteristics of the analysed waveform to best effect. EW

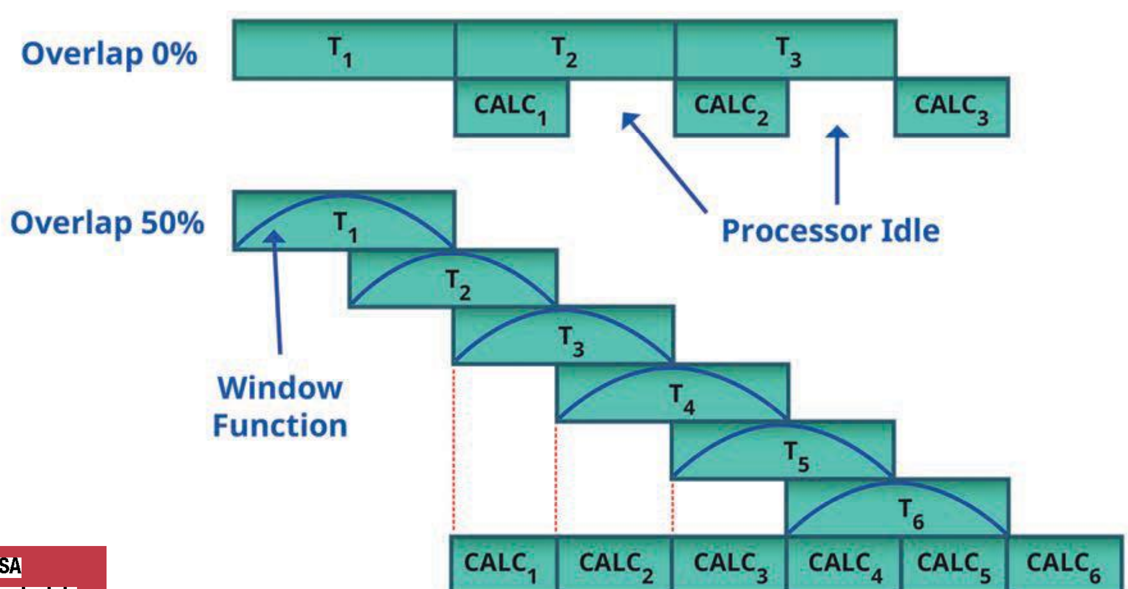


Figure 5: RTSA overlapping principle