Understanding Spectrum & Signal Analysis

Introduction

Background

Rapid recent progress in development of wireless and broadcasting technologies, such as smartphones, wireless LAN, wireless sensor networks, RFID, GPS, digital TV, etc., has seen their widespread adoption in most people's daily lives.

A variety of measuring instruments, such as frequency counters, field strength meters, power meters, etc., is used to measure, analyze, and evaluate RF signals including electromagnetic (radio) waves broadcast by antennas; the spectrum analyzer plays a central role in detailed measurement, analysis, and evaluation of RF signals. The importance of the spectrum analyzer for engineers dealing with RF signals seems unlikely to change in the future.

At the same time, progress in wireless communications technologies and the appearance of new applications have increased the complexity and level of the functions and performance required by spectrum analyzers. Among these requirements, the need to evaluate signals using new wideband digital modulation methods as well as to capture transient signal spectrums to troubleshoot problems with electromagnetic noise resulting from use of high sensitivity parts mounted at high densities requires new generations of spectrum analyzers using frequency sweep principles.

As a result, the spectrum analyzer continues to evolve to meet the needs of engineers. As an example, the digitization of internal processing has led to huge jumps in measurement speed, accuracy and stability.

As devices and computation speeds have become increasingly faster and more accurate, Anritsu has been incorporating developments in digital processing as signal analyzer functions into its MS2830A and MS2690A series.

Such improvements not only support evaluation and analysis of digital modulation signals, but also support capture of various signals with non-regular spectrums.

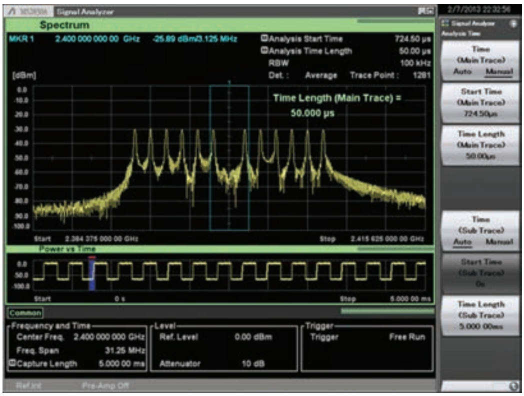


Figure 1

Engineers and technicians involved in modern RF or microwave communications have many measuring instruments at their disposal, each designed for specific measurement tasks. Among those available are:

a) **The Oscilloscope** – primarily developed for measuring and analyzing signal amplitudes in the time domain. (Voltage vs. time) Often 2, 4 or more channels of voltage vs. time can be viewed on the same display to show the relationships between signals. Extensive methods to trigger signals are often available to capture and display rare events.

b) **The Spectrum Analyzer** – designed to measure the frequency and amplitude of electromagnetic signals in the frequency domain. (Amplitude vs. Frequency & Amplitude vs. time) Most modern analyzers also have the capability to demodulate analog modulated signals. Spectrum analyzers are the most versatile tools available to the RF engineer. This guide will describe the critical performance characteristics of spectrum and signal analyzers, the types of signals measured, and the measurements performed

c) **The Signal Analyzer** – invaluable for measuring the modulation characteristics of complex signals. These units capture and process blocks of spectrum to reveal amplitude and phase relationships between signals. Newer models provide demodulation of digitally modulated signals used in most of today's communications systems.

d) **The Signal Generator** – an essential item of equipment for any communications test laboratory or workshop. The cost of a signal generator largely depends on the additional functions and facilities available as well as the type and quality of the frequency reference used.

e) **The Field Strength Meter (F.S.M.)** – display the power density of an electrical signal incident on a calibrated antenna and thus give a direct reading of field strength in dBμV/m.

f) **The Frequency Counter** – a digitally based instrument that measures and displays the frequency of incoming signals. Some models can also count 'pulse' and 'burst' signals.

Frequency Domain / Time Domain

As mentioned in the introduction, electromagnetic signals can be displayed either in the time domain, by an oscilloscope, or in the frequency domain using a spectrum or signal analyzer. Traditionally, the time domain is used to recover the relative timing and phase information required to characterize electrical circuit behavior. Many circuit elements such as amplifiers, modulators, filters, mixers and oscillators are better characterized by their frequency response information. This frequency information is best obtained by analysis in the frequency domain. Modern Oscilloscopes provide frequency domain display modes and modern Spectrum and Signal analyzers provide time domain displays. One key difference between oscilloscopes and spectrum/signal analyzers is the resolution of the vertical axis. Oscilloscopes provide high resolution along the time axis but low (8 bit) amplitude resolution. Spectrum and signal analyzers provide high (16 bit or more) amplitude resolution to see small signals in the presence of large signals.

In order to visualize these 'domains' refer to Figure 2.

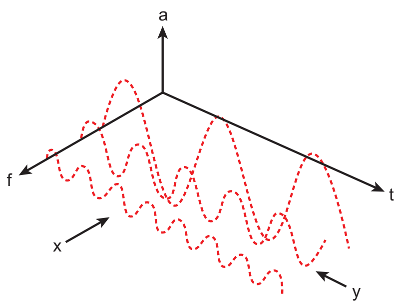


Figure 2

This represents an electromagnetic signal as a 3 dimensional model using:

(i) a time axis (t)
(ii) a frequency axis (f) and
(iii) an amplitude axis (a)

Observing from position X produces an amplitude time display where the resultant trace is the sum of the amplitudes of each signal present. This time domain view facilitates analysis of complex signals, but provides no information on the individual signal components (Figure 3).

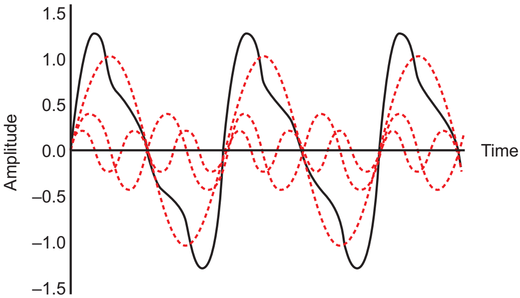


Figure 3

Viewing the model in Figure 2 from position Y, however, produces an amplitude vs. frequency display showing each component of the signal in the complex waveform. Observation in this frequency domain permits a quantitative measurement of the frequency response, spurious components and distortion of circuit elements (Figure 4).

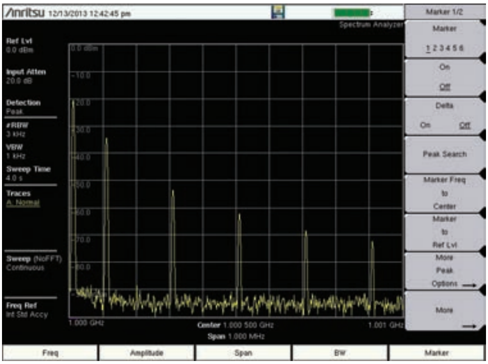


Figure 4

Spectrum Analyzers

A Spectrum Analyzer is a swept tuned analyzer. It is tuned by electronically sweeping its input over the desired frequency range thus, the frequency components of a signal are sampled sequentially in time (Figure 5). Using a swept tuned system enables periodic and random signals to be displayed but does not allow for transient responses.

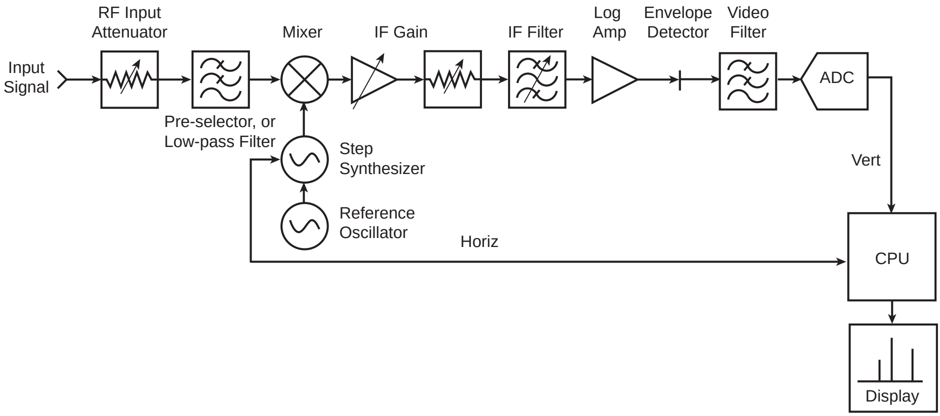


Figure 5

Signal Analyzers

Signal analyzers sample a range of frequencies simultaneously, thus preserving the time dependency and phase between signals. The signal analyzer makes use of high-level digital processing technologies, such as high-speed A/D conversion, FFT processing, digital filters, etc., to capture a spectrum that changes over time, such as for a burst signal, as well as a spectrum with transient variations. Signal Analyzers and Spectrum analyzers have very similar RF block diagrams, differing in frequency range (bandwidth) of the IF processing. The high bandwidth processing offers many advantages, but at increased cost.

System

To make best use of the signal analyzer functions and performance, it is important to understand how the signal analyzer captures the spectrum.

Figure 6 shows the signal processing flow and operation principles of a signal analyzer.

Signal analyzers capture the spectrum results using three main processes: frequency conversion, digitization and storage, and conversion to the spectrum.

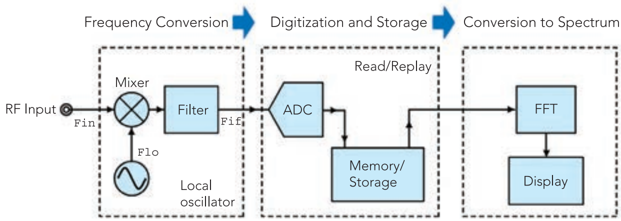


Figure 6: Signal analyzer Principle and Operation

First, the measurement signal input in a spectrum analyzer is converted to an intermediate frequency (IF) by the frequency conversion section composed of a local oscillator, mixer and bandpass filter.

The intermediate frequency `Fif` is found from the following equation where `Fin` is the measured signal frequency and `Flo` is the local oscillator oscillation frequency.

$$Fif = Fin - Flo \qquad \text{(Eq 3-1)}$$

In a signal analyzer, the value of Flo is fixed during measurement. In addition, the IF has a wide bandwidth corresponding to the measurement bandwidth (frequency span). As a result, the measured signal spectrum is frequency-converted to IF without changing the shape at Fin.

Next, the IF-converted measured signal is converted to digital data using an A/D converter.

In other words, the signal analyzer captures the true form of the measured signal in a fixed time period just by frequency conversion. The digitized time series waveform data is immediately captured to the internal memory and this data can 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 analyzed using digital processing. Deploying digital processing using Fast Fourier Transform (FFT) in the frequency domain captures the spectrum in the read time range.

The data stored in memory or hard disk is the basis of the signal analyzer 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 analyzed repeatedly using different methods and settings. This is a key feature of signal analyzers that is not supported by sweep-type spectrum analyzers.

For example, the measurement results can be confirmed using saved data by changing to a different RBW at measurement.

Figure 7: Sequential display of wireless signal spectrum display captured in memory

Operation of each part

Input and frequency conversion

In an actual analyzer, an attenuator (signal attenuator) and preamplifier in the input section assure the correct measured signal level.

At frequency conversion, although it is possible to obtain the IF signal using a single conversion, in measuring instruments, the final IF signal is obtained using multiple frequency conversions rather than a single conversion. In addition, if the frequency of the measured signal is higher than the measured signal frequency, sometimes a frequency converter is used at the input stage. The signal path until frequency conversion is the same in a signal analyzer and spectrum analyzer and the items to consider are the same.

Noise and non-linearity generated in the analog section from input to IF conversion is added to the measured value and is part of an analyzer's basic performance. Consequently, high-level analog circuit technology is essential when handling very small high-frequency signals.

As a result, in the MS2830A for example, a very low displayed average noise level of –150.5 dBm/Hz is assured to permit wide dynamic range measurements.

Signal Analyzer MS2830A specifications:

Displayed Average Noise Level (DANL): –150.5 dBm/Hz (Signal analyzer mode, 30 MHz ≤ Frequency < 1 GHz, Pre-amp: Off)

Local oscillator

The local signal oscillator handles frequency modulation of the measured signal. In a spectrum analyzer, the local oscillator oscillation frequency is swept (changed continuously) at measurement, but in a signal analyzer it oscillates at a fixed frequency during measurement.

In the same way that noise and non-linearity generated in the analog section from input to IF conversion form the basis of an analyzer's basic performance, the signal purity of the local oscillator is directly related to basic performance. For example, if there is even a slight variation in the oscillation frequency, the measured signal will appear to fluctuate at that moment too.

Similarly, short-term variations will cause Single Side Band (SSB) phase noise and widespread errors in the spectrum tails.

Consequently, the local oscillator section of a signal analyzer incorporates a high-purity oscillator.

Figure 8 shows an example of signal analyzer SSB phase noise characteristics (MS2830A Option-066).

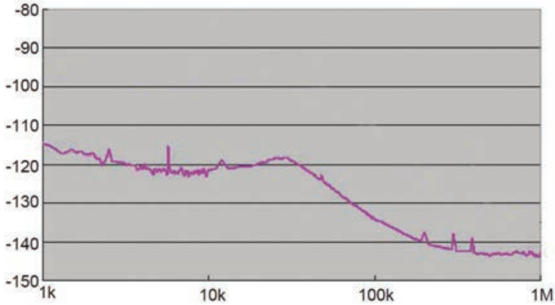


Figure 8: MS2830A Option - 066 SSB phase noise characteristics

IF circuit

The IF-converted measured signal is band-limited by the bandpass filter while being simultaneously amplified to the level required for analysis. The signal analyzer IF requires a frequency bandwidth that is equal to or greater than the analysis frequency width (span).

Consequently, the signal analyzer IF circuit is designed to achieve flat characteristics over a wide band of about 30 MHz normally, and sometimes exceeding 100 MHz, depending on the situation.

A/D converter

The frequency-converted IF (analog) measured signal is converted to a digital (time-series data) signal by the A/D converter. At A/D conversion, the signal analyzer IF requires high-speed conversion from the wideband signal.

In addition, supporting wide-dynamic-range analysis requires high-resolution A/D conversion. Consequently, the signal analyzer has a built-in high-speed, high-resolution A/D converter.

Digital processing

The digitized measured signal is saved to memory. In addition, it can be either saved to internal hard disk or transferred to an external storage device for later loading and replay.

At this time, the read data is converted to the necessary narrow range by FFT and repeated data loading and FFT creates a spectrum series changing with time which is displayed on-screen to give the impression of a spectrum changing dynamically in real time (Figure 9).

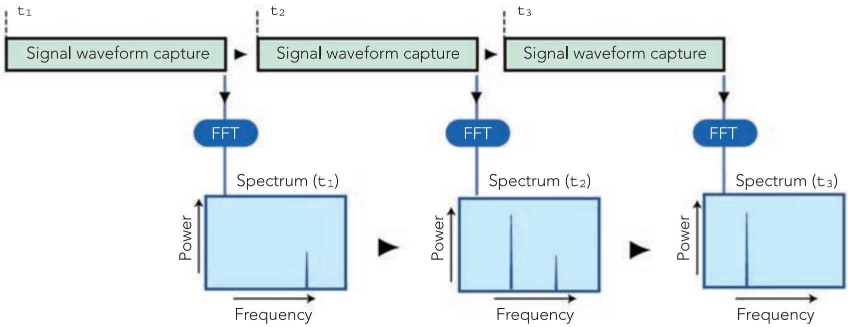


Figure 9: Continuous analysis at short cycle

Moreover, as shown in Figure 10, after saving long-term data, loading the data by slightly shifting the analysis time range and performing FFT makes it possible to sequentially inspect spectrum transitions with time at any point and any speed.

Additionally, besides performing FFT on read data to obtain the spectrum, it is also possible to obtain other information such as changes in power with time by performing processing on read data using other parameters.

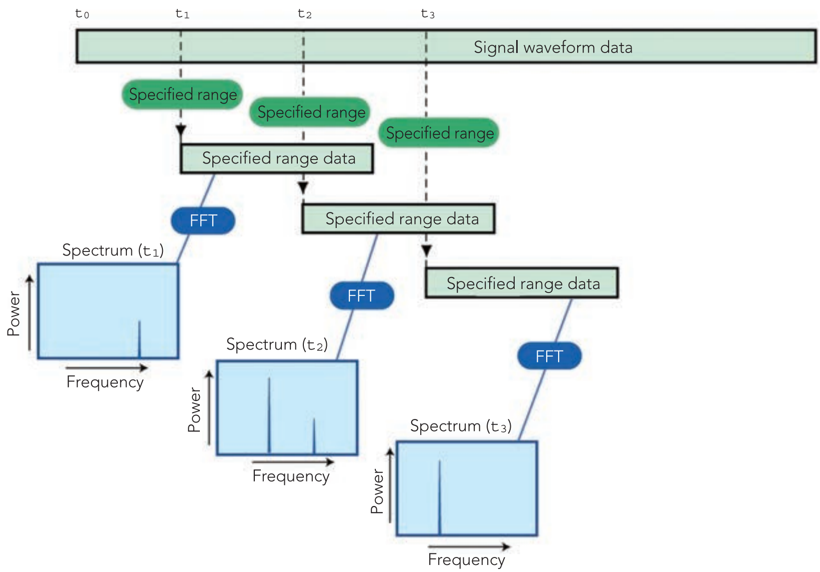


Figure10: Analysis of specific signal range recorded in memory

Sampling rate and sample count

The A/D-converted digital IF signal is resolved into the I-Q components (In-phase/Quadrature-phase) after waveform correction, etc., and is stored in memory following various procedures such as decimation, floating point data processing, etc.

Rather than leaving the data as a time-series waveform (scalar quantity), resolving and saving as I-Q data maintains the signal phase data as a vector quantity, which can assure the flexibility for post-processing, such as digital modulation analysis, etc.

The cumulative time range (capture time) can be saved to memory as data counts (samples) allocated by capture period (sampling rate).

In an actual signal analyzer, the standard memory capacity is 1 Gbyte, supporting a maximum data storage capacity of about 100 Msamples per measurement.

The sampling rate is limited by the frequency span at FFT processing, but in a signal analyzer the sampling rate is selected automatically by setting the analysis frequency span.

Consequently, there is no necessity to be concerned about sampling rate at usage. The max. capture time and max. sample counts are decided at the same time as the sampling rate, but usually rather than capturing the max. sample count, the optimum sample count is set for the measurement conditions.

Table 1 lists the sampling rates for the set frequency span and the relationship between the max. capture time and max. sample count.

Frequency span	Sampling rate	Max. capture time	Max. sample count
1 kHz	2 kHz	2000 s	4 M
10 kHz	20 kHz	2000 s	40 M
100 kHz	200 kHz	500 s	100 M
1 MHz	2 MHz	50 s	100 M
10 MHz	20 MHz	5 s	100 M
31.25 MHz	50 MHz	2 s	100 M

Table 1: Frequency span, Sampling rate, Max. capture time and Max. sample count (Extract for MS2830A)

FFT (Fast Fourier Transform)

Any time range of the I-Q data saved in memory can be read. The read data is FFT processed to generate the spectrum for the relevant time range. Similarly, processing with other parameters can also generate other signal displays, such as Power vs. Time, etc.

The spectrum obtained by FFT processing is a sequence of amplitude values for discrete frequency points. In addition, the frequency span obtained by FFT depends on the sampling frequency while the frequency gap is inversely proportional to the number of data points (sample count). Consequently, the data used for FFT is digitized at the sampling rate that satisfies the frequency span for spectrum display while also assuring a sample count supporting the frequency resolution. Put another way, the data for FFT must be digitized as the "sampling frequency" and "sample count" that achieves the expected "frequency span (frequency range)" and "resolution" at spectrum display. Incidentally, in a spectrum analyzer, the frequency resolution is evaluated as the Resolution Bandwidth (RBW); the time required for measurement, or the sweep time, is determined by the RBW and frequency span.

A signal analyzer emulates the usability of a spectrum analyzer and uses the same concept of RBW as a spectrum analyzer to improve the consistency of data measured using a spectrum analyzer. To start with, the sampling rate and data length required for FFT processing supporting the frequency span and RBW settings are set automatically.

Additionally, the RBW corresponding to the frequency span can be set automatically. At this time, in the same way as a spectrum analyzer, the measurement time increases because the number of calculations increases as the RBW becomes smaller (frequency resolution increases).

RBW / Span	1 kHz	3 kHz	10 kHz	30 kHz	100 kHz
31.25 MHz	262144	65536	32768	8192	2048
10 MHz	131072	32768	8192	4096	2048
1 MHz	8192	4096	2048	2048	2048

MS2830A Spectrum trace marker result: Integration

Table 2: Frequency span, RBW and Data points (Extract)

FFT processes the time range to be analyzed as one period and performs the same continuous repetitions on the signals before and after (the time range).

As a result, a difference in the values near both sides of the analysis time range causes an error in which the spectrum appears to widen, etc. To prevent this, the signal analyzer performs window function processing using a Gaussian window, but this requires there to be data before and after the analysis time range (Figure 11).

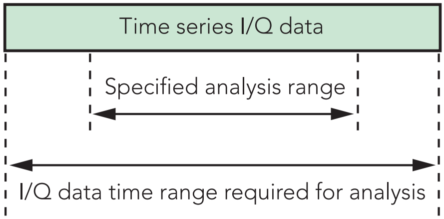


Figure 11: Required data range at analysis

As a result, the data length used for the analysis calculations (I-Q data length) becomes longer than the target analysis time (width). This also happens with non-FFT analysis (trace) modes, For example, Power vs. Time plots require data before and after the analysis time length to perform processing for the detection and moving average, and for the spectrogram window function and detection.

The length of the required before and after data differs with the trace type and the marker measurement result read accuracy setting (Marker result). Usually, the minimum data length required to capture the results as quickly as possible is set automatically (Capture time: Auto).

Obviously, analysis can be performed using the maximum memory by setting any time for the captured data length (Capture time: Manual).

In either case, the analysis range and the data range used for calculation are expressed as color bands on the time axis (Figure 12).

Figure 12: Data range display

Traces

The signal analyzer digitizes the signal and then uses FFT to capture the instantaneous spectrum, but a spectrum is not the only method for displaying a changing signal. A signal analyzer also has several trace display methods using captured data to measure and evaluate 3D signals from various viewpoints.

Spectrum trace

The Spectrum trace displays the basic spectrum with frequency on the horizontal axis and power on the vertical axis.

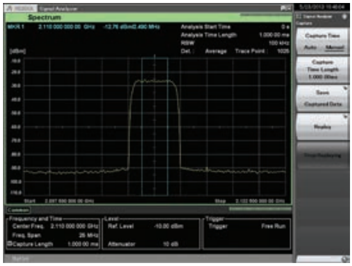


Figure 13: Example of spectrum trace (W-CDMA signal

Power vs. Time trace

The Power vs. Time trace is a method for displaying and observing changes in signal power with time. It is good for understanding signals that switch On/Off with time and amplitude variations.

The horizontal axis is time and the vertical axis is power.

Figure 14: Example of Power vs. Time trace (ARIB T61 Emergency radio burst signal

Frequency vs. Time trace

The Frequency vs. Time trace can be used to grasp signal frequency drift and sudden frequency fluctuations. The horizontal axis is time and the vertical axis is frequency.

Figure 15: Example of Frequency vs. Time trace (Frequency deviation of FM waveform)

Phase vs. Time trace

The Phase vs. Time trace is good for analysis of signal phase time stability and phase modulation, etc. The horizontal axis is time and the vertical axis is phase.

Main Traces

Figure 16: Example of Phase vs. Time trace (Phase characteristics of 1.9 GHz band unmodulated signal)

CCDF trace

The CCDF trace is useful for comparing the digital modulation signal peak and average as well as for confirming the amplitude distribution of non-periodic noise in a signal. The horizontal axis is amplitude and the vertical axis indicates the probability that the signal is at each amplitude. The probability is displayed as CCDF or PDF.

CCDF (Complementary Cumulative Distribution Function):

Cumulative distribution of instantaneous power deviation vs. average power

APD (Amplitude Probability Distribution):

Probability distribution of instantaneous power vs. average power

Figure 17: Example of CCDF trace (CCDF of terrestrial digital TV signal)

Spectrogram trace

The Spectrogram trace is the equivalent of looking down from the top of the power axis at the power, frequency and time data represented in 3D (Figure 19) to get a good overall grasp of the signal spectrum and power changes.

The horizontal axis is time and the vertical axis is frequency.

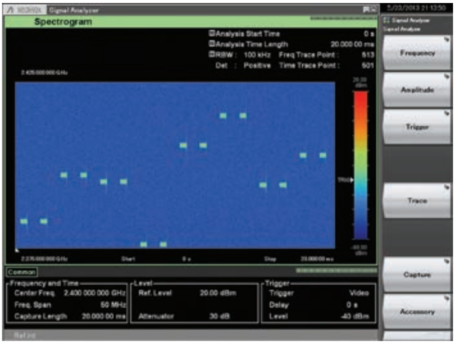


Figure 18: Example of Spectrogram trace (Frequency hopping of Bluetooth signal)

The power level is color-coded.

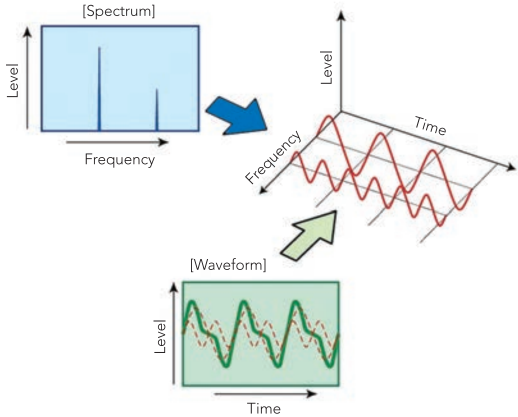


Figure 19: Signal observation direction

Sub-traces (Split screen display)

This function displays a sub-trace under the main trace as a supplement to the normal main trace.

The sub-trace can be selected to display either the Power vs. Time or Spectrogram trace with the trace displaying data for any time period. When the sub-trace is displayed, the main trace results are conveniently displayed immediately above for the corresponding point in time. Additionally, setting the analysis range at the sub-trace and slowly shifting the analysis start point slightly using the main dial displays a time-lapse animation of the spectrum on the main trace.

Figure 20: Example of split screen display (Main trace: Spectrum, Sub-trace: Power vs. Timel)

Basic Operation

Both spectrum analyzers and signal analyzers are based on a super heterodyne receiver principle (Figure 21). The input signal, fIN, is converted to an intermediate frequency, fIF, via a mixer and a tunable local oscillator fLO. When the frequency difference between the input signal and the local oscillator is equal to the intermediate frequency then there is a response on the display.

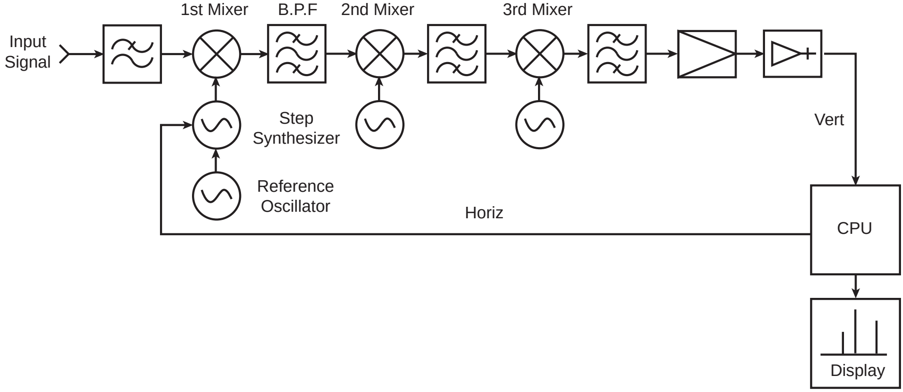


Figure 21

$$f_{IN} = f_{LO} \pm f_{IF}$$

This is the basic tuning equation that determines the frequency range of a spectrum/signal analyzer. Using the super heterodyne technique enables high sensitivity through the use of intermediate frequency (IF) amplifiers and extended frequency range by using the harmonics of the local oscillator (LO). This technique is not, however, real time and sweep rates must be consistent with the IF filter bandwidth charge time.

Characteristics

Spectrum and signal analyzers have the following characteristics:

- Wide frequency range.
- Amplitude and frequency calibration via internal calibration source and error correction routines.
- Flat frequency response where amplitude is independent of frequency.
- Good frequency stability using synthesized local oscillators and reference source.
- Low internal distortion.
- Good frequency resolution.
- High amplitude sensitivity.
- Linear and logarithmic display modes for amplitude (voltage and dB scaling).
- Absolute and relative measurement capabilities.

Frequency Range

The lower frequency limit of a spectrum analyzer is determined by the sideband noise of the local oscillator. The local oscillator feedthrough occurs even when there is no input signal present.

The sensitivity at the lower frequency is also limited by the LO. sideband noise. Figure 22 shows typical data of average noise level vs. frequency for two IF bandwidths.

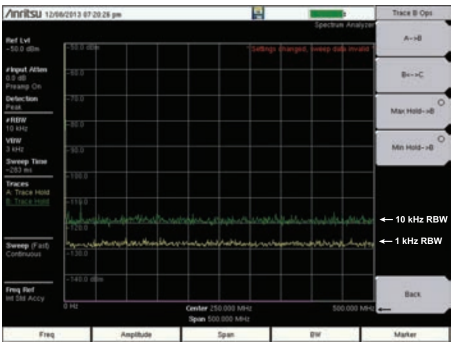


Figure 22

It should be noted however, that as the IF bandwidth is reduced so the time to sweep a given frequency range increases since the charge time of the IF filter increases. This means that the sweep time is increased to allow the IF filter to respond and therefore present an undistorted signal to the detector. These variables are generally taken into account automatically and are referred to as 'coupling'. Beyond the detector can be more filtering known as Video Bandwidth and this can also be coupled to IF bandwidth and sweep time. These functions are coupled together since they are all inter dependent on each other, i.e. change one parameter setting and it affects the others.

An additional facility available on most modern analyzers is a Zero Frequency Span mode. As mentioned earlier, most analyzers are based on the super heterodyne receiver design, where the local oscillator is swept continuously. If the local oscillator is manually tuned, the spectrum analyzer becomes a fixed tuned receiver whose frequency is determined by that of the local oscillator. In this mode the analyzer will display the time domain function since the frequency component is fixed even though the scan generator is still sweeping the display i.e. the display is now amplitude vs. time (Figure 23).

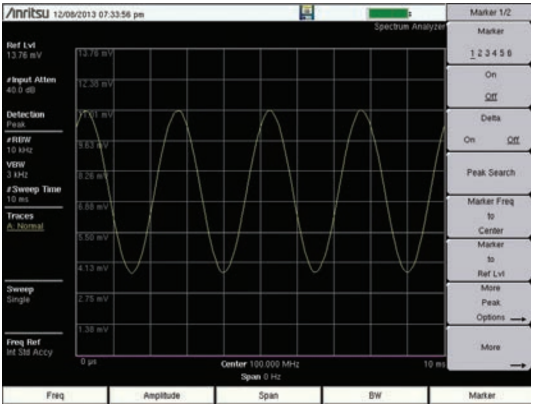


Figure 23

Frequency Resolution

The frequency resolution (typically called "resolution bandwidth") of a spectrum/signal analyzer is its ability to separate and measure two signals in close proximity. This frequency resolution is determined by three primary factors:

- the IF filter bandwidth used
- the shape of the IF filter and
- the sideband noise of the IF filter

The IF bandwidth is normally specified by Δf at 3 dB (Figure 24). From this it can be seen that the narrower the filter bandwidth the greater the frequency resolution. However, as mentioned earlier, as the IF band width is reduced so the charge time for the filter increases hence increasing the sweep time. As an ex ample, narrow IF bandwidths are required to distinguish the sidebands of amplitude and frequency modulated signals (Figure 25).

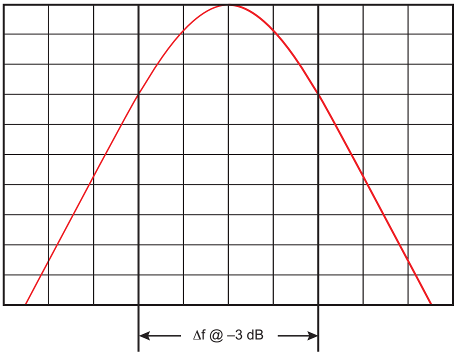


Figure 24

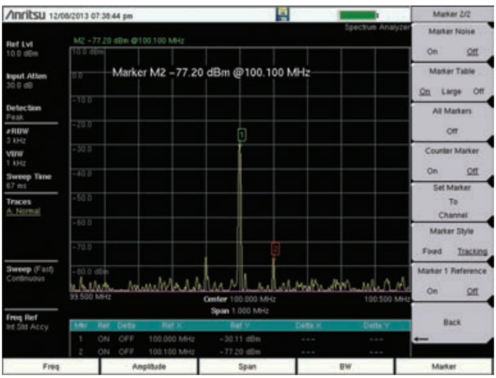


Figure 25

When measuring close in spurious components, the shape of the IF filter becomes important. The filter skirt inclination is determined by the ratio of the filter bandwidth at –60 dB to that at –3 dB (Figure 26).

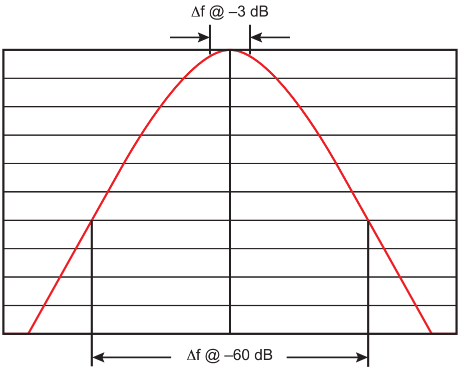


Figure 26

This skirt inclination is known as the 'shape factor' of the filter and provides a convenient guide to the filter quality. The most common type of IF filter is known as the Gaussian filter, since its shape can be de rived from the Gaussian function of distribution. Typical shape factor values for Gaussian filters are 12:1/ 60 dB:3 dB, while some spectrum analyzers utilize digital filters where the shape factor can be as low as 3:1. Digital filters appear to be better in terms of frequency resolution, but they do have the drawback of sharply increasing the scan time required to sweep a given frequency range. Figure 27 shows the effects of scanning too fast for a given IF bandwidth filter. As the scan time decreases, the displayed amplitude decreases and the apparent bandwidth increases. Consequently, frequency resolution and amplitude uncertainty get worse, and some analyzers will warn you that you are now in an 'UNCAL' mode.

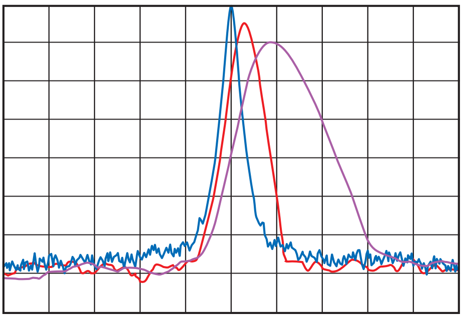


Figure 27

A spectrum analyzer's ability to resolve two closely spaced signals of unequal amplitude is not only dependent on the IF filter shape factor. Noise sidebands can reduce the resolution capabilities since they will appear above the skirt of the filter and so reduce the out of band rejection of the filter.

Sweep Speed

Spectrum analyzers, incorporating swept local oscillators have the issue of needing to manage the sweep speed to prevent uncalibrated displays. Signal analyzers do not. Blocks of spectrum are processed together in a signal analyzer. See Figure 6. The sample rate of the A to D converter determines the span of spectrum that can be processed. The span is approximately ½ the A to D sample rate. The difference in sweep speed performance between the spectrum analyzer mode and signal analyzer mode is especially visible when very narrow resolution bandwidths are used. Most modern analyzers combine swept spectrum and signal analyzer technology. The spectrum analyzer mode offers very wide span views and the signal analyzer mode offers fast spectrum displays for narrow spans. The sweep speed for signal analyzer-based spectrum displays depends on the FFT computation speed. Dedicated FFT processing circuitry can speed up spectrum display rates to support searching for intermittent signals.

Sensitivity and Noise Figure

The sensitivity of a spectrum analyzer is defined as its ability to detect signals of low amplitude. The maximum sensitivity of the analyzer is limited by the noise generated internally. This noise consists of thermal (or Johnson) and non-thermal noise. Thermal noise power is expressed by the following equation:

$$P_N = kTB$$

where

PN = Noise power (in Watts)

k = Boltzman's constant (1.38 x 10^{23} JK^{-1})

T = Absolute temperature (Kelvin)

B = System Bandwidth (Hz)

From this equation it can be seen that the noise level is directly proportional to the system bandwidth. Therefore, by decreasing the bandwidth by an order of 10 dB the system noise floor is also decreased by 10 dB (Figure 28).

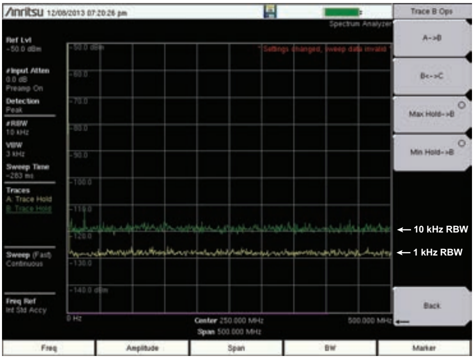


Figure 28

When comparing spectrum analyzer specifications it is important that sensitivity is compared for equal bandwidths since noise varies with bandwidth.

An alternative measure of sensitivity is the noise factor FN:

$$F_N = (S/N)_{IN} \,/\, (S/N)_{OUT}$$

where S = Signal and N = Noise

Since the noise factor is a dimensionless figure of merit we can derive the noise figure as:

$$F = 10 \log (F_N) \text{ dB}$$

Using the equation PN = kTB it is possible to calculate the theoretical value of absolute sensitivity for a given bandwidth. For example, if a spectrum analyzer generates no noise products at a temperature of 17 degrees Celsius, referred to a 1Hz bandwidth, then:

absolute sensitivity = $1.38 \times 10^{-23} \times 290$

= 4×10^{21} W/Hz

= –174 dBm/Hz

To determine the noise figure of a typical spectrum analyzer where the average noise floor is specified as 120 dBm referred to a 300 Hz bandwidth:

–120 dBm = –174 dBm/Hz + 10 log 300 + F (dB)

F (dB) = –120 + 174 –24.8

Noise Figure = 29.2 dB

Video Filtering or Averaging

Very low level signals can be difficult to distinguish from the average internal noise level of many spectrum analyzers. Since analyzers display signal plus noise, some form of averaging or filtering is required to assist the visual detection process. As mentioned earlier, a video filter is a low pass, post detection filter that averages the internal noise of the analyzer.

Because spectrum analyzers measure signal plus noise, the minimum signal power that can be displayed is the same as the average noise power of the analyzer. From this statement it would appear that the signal would be lost in the analyzer noise but:

if signal power = average noise power

then by definition, the minimum signal power that can be displayed will be:

$$\frac{S + N}{N} = 2$$

Where

S = signal power

N = average noise power

When the signal power is added to the average noise power, the resultant signal power displayed will be 3 dB greater (Figure 29). This 3 dB difference is sufficient for low level signal identification.

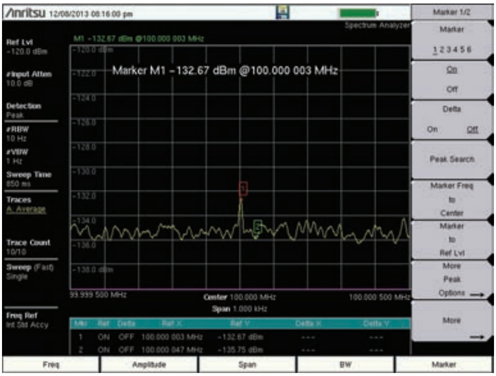


Figure 29

Signal Display Range

The signal display range of a spectrum/signal analyzer with no input attenuation is dependent on two key parameters.

- The minimum resolution bandwidth available and hence the average noise level of the analyzer and
- The maximum level delivered to the first mixer that does not introduce distortion or inflict permanent damage to the mixer performance.

Typical values for these two factors are shown in Figure 30.

As the input level to the first mixer increases so the detected output from the mixer will increase. However, since the mixer is a semiconductor diode the conversion of input level to output level is constant until saturation occurs. At this point the mixer begins to gain compress the input signal, and conversion reverts from linear to near logarithmic. This gain compression is not considered serious until it reaches 1 dB.

Input levels that result in less than 1 dB gain compression are called linear input levels (Figure 31). Above 1 dB gain compression, the conversion law no longer applies and the analyzer is considered to be operating nonlinearly and the displayed signal amplitude is not an accurate measure of the input signal.

Distortion products are produced in the analyzer whenever a signal is applied to the input. These distortion products are usually produced by the inherent nonlinearity of the mixer. By biasing the mixer at an optimum level internal distortion products can be kept to a minimum. Typically, modern spectrum analyzer mixers are specified as having an 80 dB spurious free measurement range for an input level of –30 dBm. Obviously the analyzer will be subjected to input signals greater than –30 dBm and to prevent exceeding the 1 dB compression point, an attenuator is positioned between the analyzer input and the first mixer. The attenuator automatically adjusts the input signal to provide the –30 dBm optimum level.

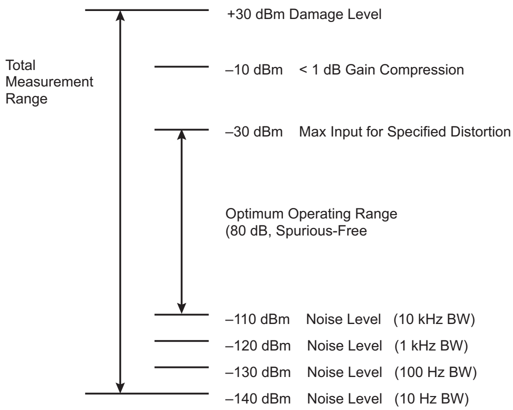


Figure 30

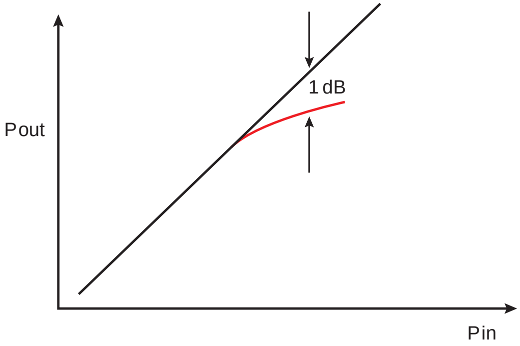


Figure 31

Dynamic Range

The dynamic range of a spectrum/signal analyzer is determined by four key factors.

- Average noise level.
  This is the noise generated within the spectrum analyzer RF section, and is distributed equally across the entire frequency range.
- Residual spurious components.
  The harmonics of various signals are mixed together in complex form and converted to the IF signal components which are displayed as a response on the display. Consequently, the displayed response is present regardless of whether or not a signal is present at the input.
- Distortion due to higher order harmonics.
  When the input signal level is high, spurious images of the input signal harmonics are generated due to the nonlinearity of the mixer conversion.
- Distortion due to two signal 3rd order intermodulation products.
  When two adjacent signals at high power are input to a spectrum/signal analyzer, intermodulation occurs in the mixer paths. Spurious signals, separated by the frequency difference of the input signals are generated above and below the input signals.

The level range over which measurements can be performed without interference from any of these factors is the dynamic range. This represents the analyzers performance and is not connected with the dis play (or measurement) range. The four parameters that determine dynamic range can normally be found in the analyzer specifications.

For simplicity, some analyzer specifications state the dynamic range as “Y dB for an input level of X dBm”. The following example shows how these parameters are related to dynamic range:

Amplitude Dynamic Range: 70 dB for a mixer input signal level of –30 dBm (Atten. = 0 dB) In order to achieve this value of dynamic range the following conditions are required:

a) the IF bandwidth must be narrow enough such that the average noise level is better than –100 dBm.

b) the residual spurious components must be less than –100 dBm.

c) for an input level of 30 dBm the higher harmonic distortion must be better than –70 dB (i.e. better than –100 dBm).

Analyzer manufacturers often relate the above specifications at a particular frequency or over a range of frequencies.

Frequency Accuracy

The key parameter relating to frequency accuracy is linked to the type of reference source built into the spectrum/signal analyzer.

- Synthesized
  The analyzer local oscillator is phase locked to a very stable reference source, often temperature controlled to prevent unwanted frequency drifting. In this case, a precision crystal is often used and the overall frequency accuracy and stability, both short term and long term depend on its quality. Portable analyzers, intended for outdoor use, often have GPS receivers that can significantly improve the stability of the internal local oscillator.
- Non Synthesized
  The analyzer local oscillator operates as a stand-alone voltage controlled source

Uses of Spectrum Analyzer and Signal Analyzers

Signal analyzers incorporate a wide bandwidth digitizer in the IF to capture a time block of spectrum for analysis. Frequency, time and phase relationships of signals can be analyzed within the bandwidth and time limits of the captured spectrum. Digital modulation can be characterized in many ways not possible with a swept tuned spectrum analyzer. Figure 32 compares the block diagrams for a spectrum analyzer and signal analyzer.

Figure 33 show example time blocks of spectrum with a variety of modulations. A signal analyzer is often used to measure the characteristics of analog and digital modulation.

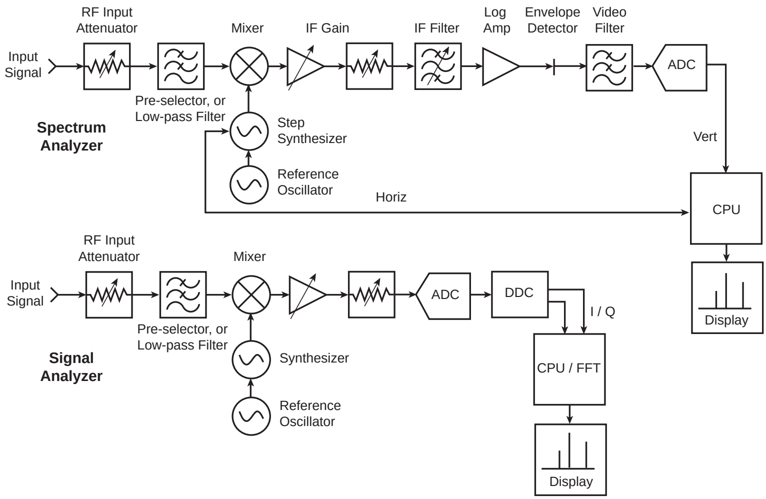
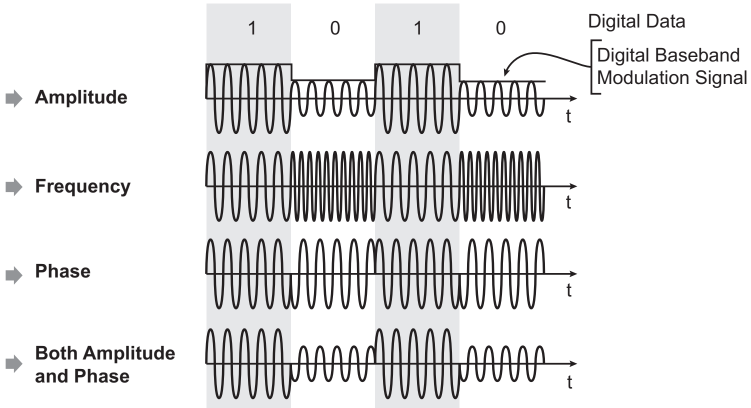


Figure 32



$U(t) = \underbrace{U(t)}_{AM} \cdot \sin(2\pi \underbrace{f_c t}_{FM} + \underbrace{\phi(t)}_{PM})$

Figure 33

Figure 34 shows a signal analyzer display of QPSK modulation in polar display format. The polar display is called a constellation or vector diagram.

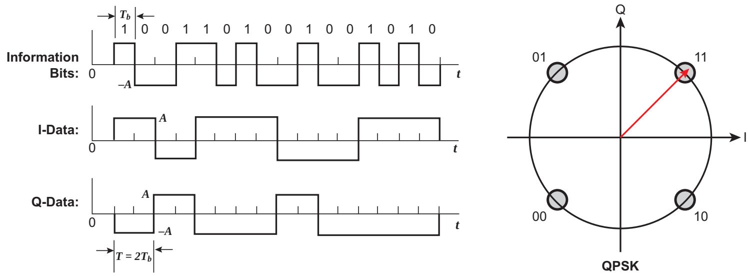


Figure 34

Structural differences and similarities

The measured signal is converted to an intermediate frequency (IF) signal at the mixer and then passed through a narrowband filter. The amplitude of the frequency in the measured signal (Fin) is determined by the detector.

Additionally, since the input frequency Fin is composed of Flo + Fif (local oscillator frequency + intermediate frequency), sweeping (continuously changing) Flo makes it possible to obtain the spectrum corresponding to the sweep range. In other words, the frequency width (span) that can be analyzed by a spectrum analyzer is determined by the sweep range of the local oscillator.

In passing, the part of Figure 35 enclosed by the dotted line (post-IF signal processing and local oscillator control) have been digitized, resulting in much faster measurement speed and higher accuracy. Figure 35 shows the operation principles of the spectrum analyzer.

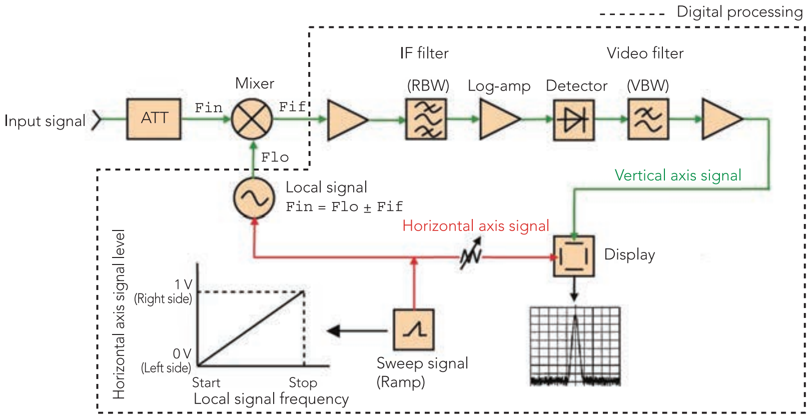


Figure 35

If we treat the digital part of the spectrum analyzer (part inside dashed line in Figure 35) as a black box, comparison with the signal analyzer operation principles shows the circuit diagram of both the spectrum analyzer and signal analyzer to be very similar. One reason is because the signal analyzer offers added spectrum analyzer functions. On the other hand, compared to the signal analyzer, which captures the spectrum using FFT, a big difference between the two is that the spectrum analyzer captures the spectrum according to the operating principles in Figure 35 using filtering and digital log conversion/detection.

Operation differences

Due to the different principles of spectrum capture, the signal analyzer is ideally suited for capturing irregular signals where the spectrum changes in the short term. This is a key functional difference compared to the spectrum analyzer, which is better for measuring stationary signals with a stable spectrum over time. In addition, the spectrum analyzer and signal analyzer have other differences as a consequence of the differences in the operation principles.

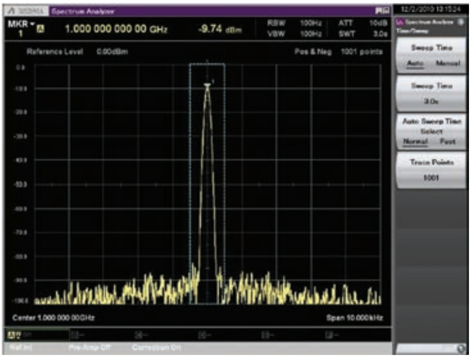
The first difference is the analysis frequency width (span). The spectrum analyzer analysis frequency width (span) is dependent on the local oscillator sweep range; the local oscillator supports sweeping of a wide frequency range exceeding 1 GHz. Conversely, the signal analyzer analysis frequency width (frequency span) is determined by the IF bandwidth. The actual IF bandwidth is several MHz to 100 MHz. As a result, the spectrum analyzer is used to measure spectrums with a wide frequency of more than several 100 MHz.

The second is the time required for measurement. The problem with spectrum analyzer measurement time is the time required for the local oscillator sweep. Consequently, sweep speed (frequency conversion rate) is limited by the frequency resolution (RBW) and a drop in the sweep speed is required to delay the response as resolution increases (RBW becomes narrower).

In comparison, the signal analyzer measurement time depends only on the time to capture and FFT-process the signal and is much shorter than the local oscillator frequency sweep. As a result, when comparing both measurement times for a relatively narrow span, the signal analyzer completes measurement and obtains the measurement in the shortest time.

The difference is striking when measurement is repeated, such as when averaging measured values (Figures 36 and 37).

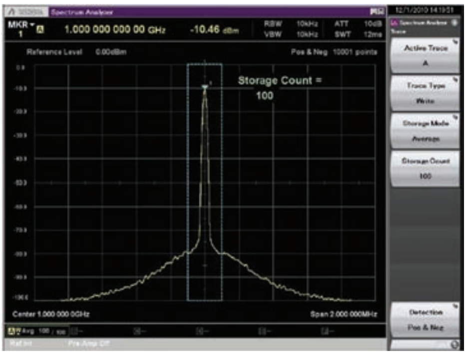
Incidentally, the effect of using averaging to decrease randomness in measured results depends on the video filter VBW setting for spectrum analyzers and on the number of measurements used at averaging processing for signal analyzers.

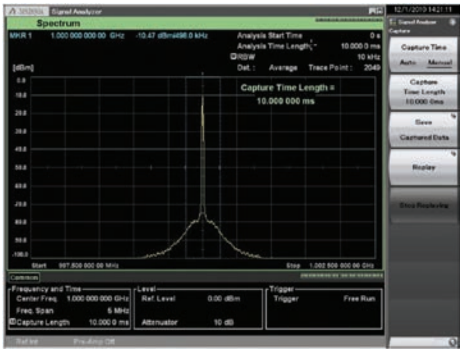


Spectrum Analyzer
Measurement time (Sweep time): 3 s
(VBW: 100 Hz)

Signal Analyzer
Capture time: 50 µs
(Average trace points: 1024)

Figure 36: Example of narrowband high resolution measurement
(Center frequency: 1 GHz, Span: 10 kHz. RBW: 100 Hz)





Spectrum Analyzer
Measurement time: 1.2 s
(Sweep time: 12 ms; Averaging count: 100, Frequency span: 2 MHz)

Signal Analyzer
Capture time: 10 ms
(Analysis time length: 10 ms, Average trace points: 2049, Frequency span: 5 MHz)

Figure 37 : Example of Averaging measurement time
(Center frequency: 1 GHz, RBW: 10 kHz)

The third is that the spectrum analyzer captures measurement results while the measured signal is connected whereas the signal analyzer uses batch processing so it can perform measurement both in real time and when offline. As a consequence, various analyses can be performed by changing the analysis method and parameters (RBW, time specifications, etc.). In addition, since the signal analyzer can record and save the measured signal to a storage device, such as a hard disk, it is possible to take data recorded on-site back to the laboratory where it can be replayed and analyzed.

Applications

As stated in the introduction, spectrum analyzers are used to display the frequency and amplitude of signals in the frequency domain. Efficient transmission of information is accomplished by a technique known as modulation. This technique transforms the information signal, usually of low frequency, to a higher carrier frequency by using a third, modulation signal. But why modulate the original signal? The two primary reasons are:

1) modulation techniques allow the simultaneous transmission of two or more low frequency, or base band signals onto a higher, carrier frequency and

2) high frequency antenna are small in physical size and more electrically efficient.

In this section we will consider three common modulation formats:

- Amplitude Modulation or AM.
- Frequency Modulation or FM.
- Pulse Modulation or PM.

Each modulation technique places emphasis on a particular area of the analyzer's specification.

Amplitude Modulation

As the name suggests, amplitude modulation is where the carrier signal amplitude is varied by an amount proportional to the amplitude of the signal wave and at the frequency of the modulation signal. The amplitude variation about the carrier is termed the modulation factor 'm'. This is usually expressed as a percentage called the percent modulation, %M.

The complex expression for an AM carrier shows that there are three signal elements.

- the unmodulated carrier.
- the upper sideband whose frequency is the sum of the carrier and the modulation frequency.
- the lower sideband whose frequency is the difference between the carrier and the modulation frequency.

The spectrum analyzer display enables accurate measurement of three key AM parameters.

- Modulation Factor m.
- Modulation Frequency fm.
- Modulation Distortion

Figure 38 shows the time domain display of a typical AM signal. From this the modulation factor, m, can be expressed as follows:

$$m = \frac{E_{max} - E_c}{E_c} \qquad \textit{Equation 1}$$

$$E_{max} - E_c = E_c - E_{min} \qquad \textit{Equation 2}$$

$$E_c = \frac{E_{max} + E_{min}}{2} \qquad \textit{Equation 3}$$

$$m = \frac{E_{max} - E_{min}}{E_{max} + E_{min}} \qquad \textit{Equation 4}$$

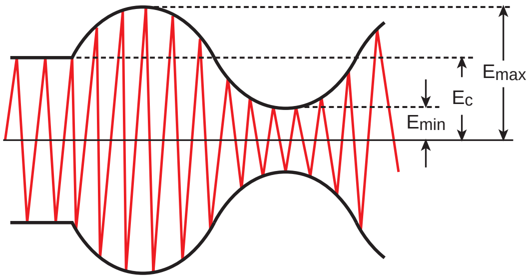


Figure 38

Equation 4 is true for sinusoidal modulation. If we view the AM signal on a spectrum analyzer in linear (voltage) mode we obtain Figure 39.

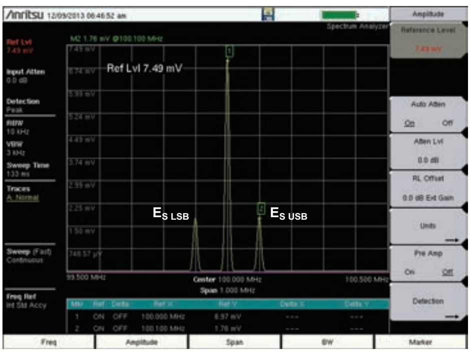


Figure 39

From this the percentage modulation, %M, can be calculated as follows:

$$\%M = \frac{(E_{S\ LSB} + E_{S\ USB})}{E_c} \times 100$$

where

Es = Amplitude of the sideband (volts)

Ec = Amplitude of the carrier (volts).

For low levels of modulation it is more convenient to use the analyzers logarithmic display as in Figure 40.

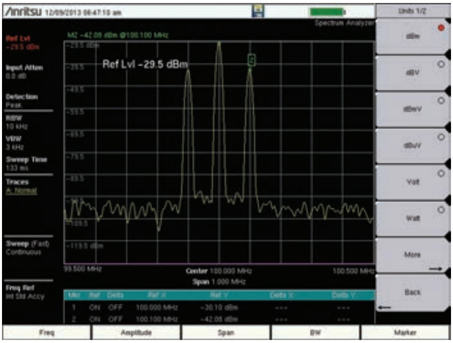


Figure 40

The relationship between the sideband level and the percentage modulation is shown in Table 3.

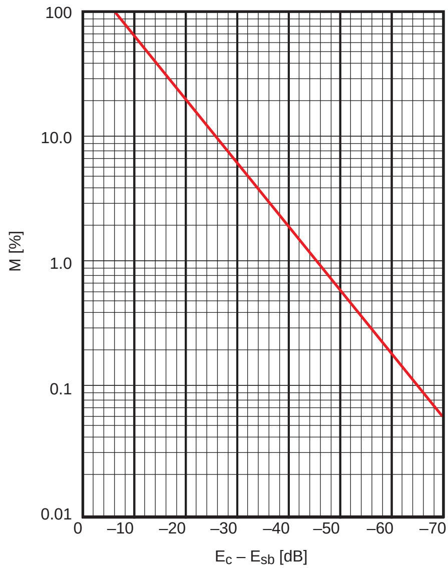


Table 3

As an example, consider a case in which the carrier frequency Fc = 1000 MHz, and the modulation frequency fm = 1 kHz

Figure 41 shows the result of observation using an oscilloscope. From the envelope, %M = 50% (m = 0.5).

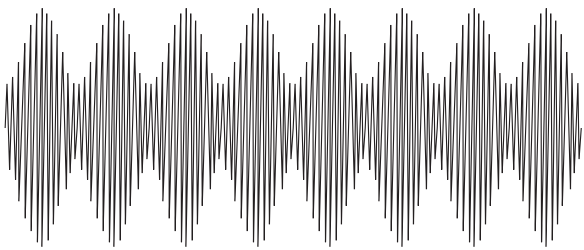


Figure 41

Figure 42 shows the same signal displayed on the linear scale (voltage) of a spectrum analyzer. From equation 5.

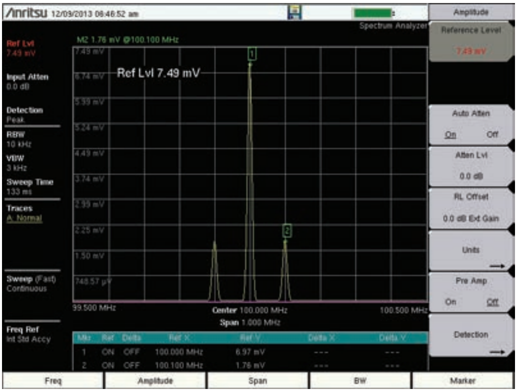


Figure 42

$$\%M = \frac{1.76\ \text{mV} + 1.76\ \text{mV}}{6.97\ \text{mV}} \times 100$$

$$\%M = 50\%$$

If m = 0.05 (%M = 5%), then for the same conditions the sideband level will be 0.165 mV for a carrier level of 6.6 mV. Clearly for low modulation factors the logarithmic display is better suited (Figure 43).

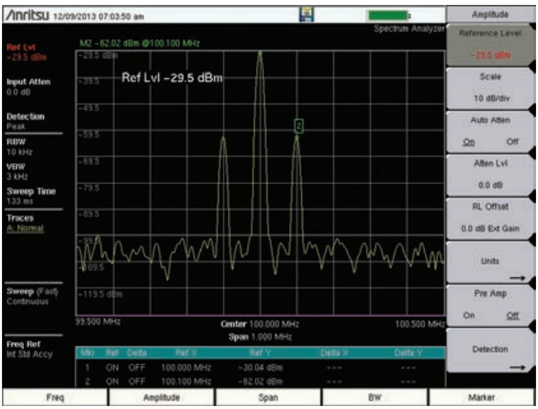


Figure 43

Modulation Frequency fm

As stated earlier, for amplitude modulation the upper and lower sidebands displayed on a spectrum analyzer will be separated from the carrier by a frequency equal to the modulation frequency (Figure 44). This frequency domain display assumes that the IF bandwidth is narrow enough to resolve the spectral components of the modulated carrier. However, a common modulation test tone of 400 Hz will be difficult to measure if the analyzer has a minimum 1 kHz resolution bandwidth. More difficulties arise if the phase noise of the carrier masks low frequency modulation sidebands with small modulation factors.

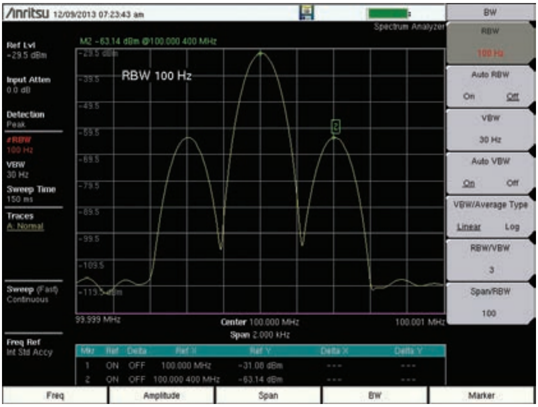


Figure 44

If the modulation factor is high enough, we can use the spectrum analyzer as a fixed tuned receiver as follows:

- Set the carrier to the center of the display.
- Ensure that the resolution bandwidth and the video bandwidth are sufficiently wide enough to en compass the modulation sidebands without attenuation.
- Select zero span and adjust the reference level so that the peak of the signal is near to the top of the screen.
- Select linear display mode, video triggering and adjust the sweep time to display several cycles of the demodulated waveform.

From this display we can measure the modulation factor, m, and the modulating frequency using the analyzers delta marker function (Figure 45).

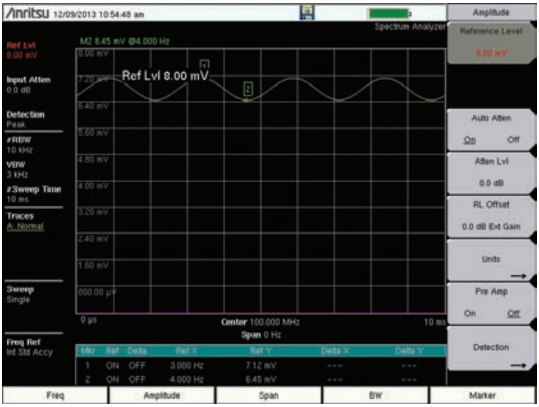


Figure 45

Note: Since this is a relative measurement, as we adjust the reference level of the analyzer, the absolute values of Emax and Emin change but the ratio remains constant. Using the delta marker function will yield the ratio E so by modifying the equation for m we can use this ratio directly.

$$m = \frac{(1 - (E_{min} / E_{max}))}{(1 + (E_{min} / E_{max}))}$$

Modulation Distortion

Distortion of an amplitude modulated carrier wave is commonly due to either or both of the following:

- second and subsequent harmonics of the modulation signal and,
- over modulation of the carrier wave. i.e. %M>100%.

Measuring modulation distortion can be performed directly from the frequency domain display of a spectrum analyzer. Consider Figure 46.

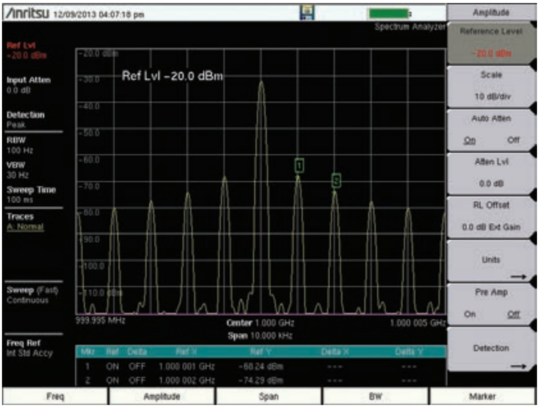


Figure 46

The upper and lower sidebands adjacent to the carrier are the modulation components but the second and subsequent pairs of sidebands are due to the harmonics of the modulation signal. Using a logarithmic scale, the level difference between the first and second sidebands gives the 2nd harmonic distortion for the waveform. In the case of Figure 46 this is 6 dB. This same procedure can be used for 3rd harmonic distortion also.

Now consider Figure 47. This shows an over-modulated 100 MHz carrier with f_m = 1 kHz. From the time domain display (Figure 48) we can see that the carrier is cut off when the modulation frequency is at a minimum. From the corresponding frequency domain display, the first sideband pair are 6 dB lower than the carrier hence %M = 100% but note also the severe harmonic distortion products.

These distortion products effectively increase the occupied bandwidth unnecessarily.

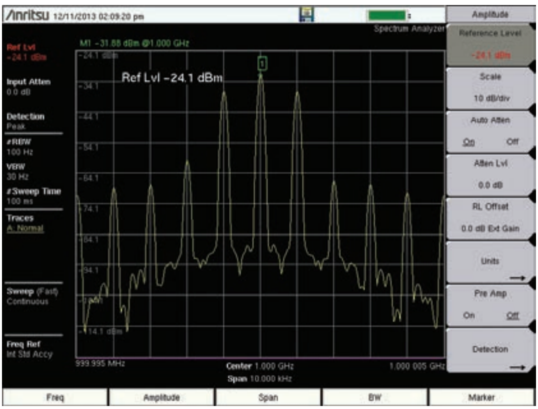


Figure 47

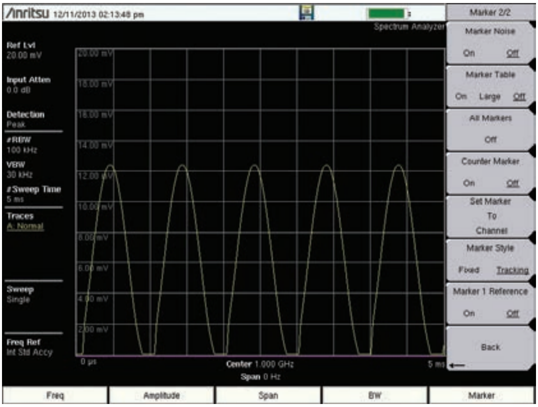


Figure 48

By definition, the information transmitted by amplitude modulation is carried not by the carrier but via the sidebands. Thus varying the composite AM waveform varies only the sideband amplitude. If the carriers component is suppressed, then the overall power saving improves the efficiency of the transmission system. This type of modulation is called Double Sideband Suppressed Carrier or DSBSC. In order to recover the modulation signal the carrier must be reinserted at the receiver.

Furthermore, we could also remove one of the sidebands since the same information is carried by both. This would result in a further power saving and a reduction in the occupied bandwidth of the signal. This type of modulation is called Single Sideband Suppressed Carrier but is usually just called Single Sideband (SSB).

Frequency Modulation

Frequency modulation, FM, is a form of modulation in which the frequency of a carrier wave is varied above and below its unmodulated value by an amount proportional to the amplitude of a signal wave and at the frequency of the modulating signal. In this case the carrier amplitude remains constant. Frequency modulation differs from amplitude modulation in a number of ways.

- Since the amplitude of the modulated carrier remains constant, regardless of the modulation frequency and amplitude, no power is added to or removed from the carrier wave of an FM signal.
- Frequency modulation of a sinusoidal carrier with a second varying sinusoid yields an infinite number of sidebands separated by the modulation frequency f_m.
- The peak-to-peak amplitude of the signal wave determines the maximum frequency deviation of the modulated carrier.

The Bessel function curves of Figure 49 show the relationship between the carrier and sideband amplitudes of a frequency modulated wave as a function of the modulation index m.

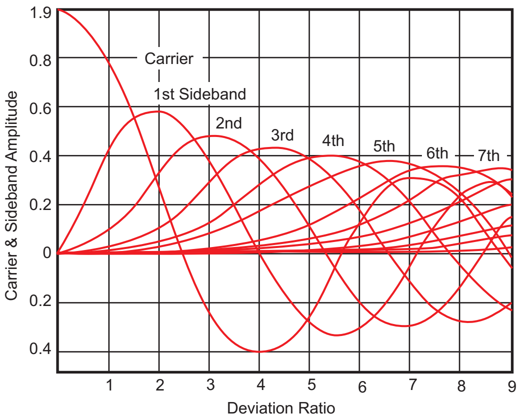


Figure 49

Note that the carrier component J_0 and the various sidebands J_N go to zero amplitude for specific values of m. From these curves we can determine the amplitude of the carrier and the sideband components in relation to the unmodulated carrier. For example, we find for a modulation index of m = 3 the following amplitudes:

Carrier $J_0 = 0.26$

First order sideband $J_1 = 0.34$

Second order sideband $J_2 = 0.49$

Third order sideband $J_3 = 0.31$

The sign of the values we get from the curves is not significant since a spectrum analyzer displays only absolute amplitudes. The exact values for the modulation index corresponding to each of the carrier zeros are listed in the Appendix C.

Bandwidth of FM Signals

In practice, the spectrum of an FM signal is not infinite. The sideband amplitudes become negligible beyond a certain frequency offset from the carrier, depending on the magnitude of m. We can determine the bandwidth required for low distortion transmission by counting the number of significant sidebands. (Significant sidebands usually refers to those sidebands that have a voltage at least 1 percent (40 dB) of that of the unmodulated carrier).

Figures 50 and 51 show the analyzer displays of two FM signals, one with m = 0.2, the other with m = 95. Two important facts emerge from these figures:

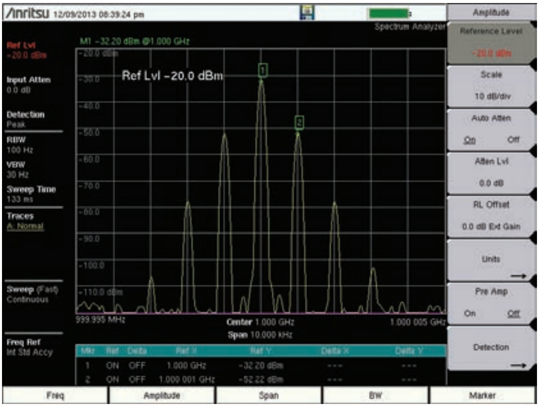


Figure 50

- For very low modulation indices (m<0.2), we get only one significant pair of sidebands. The required transmission bandwidth in this case is twice f_m, as for AM.
- For very high modulation indices (m>100), the transmission bandwidth is twice Δ fpk. For values of m between these margins we have to count the significant sidebands.

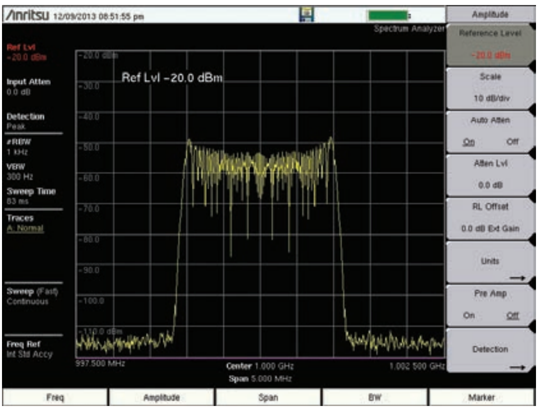


Figure 51

For voice communication a higher degree of distortion can be tolerated; that is, we can ignore all side bands with less that 10% of the carrier voltage (20 dB). We can calculate the necessary bandwidth B using the approximation:

$$B = 2\Delta F_{pk} + 2F_m$$

$$\Delta F_{pk} = m \times f_m \text{ maximum frequency deviation}$$

$$\text{or } B = 2Fm\ (1 + m)$$

So far our discussion of FM sidebands and bandwidth has been based on having a single sine wave as the modulating signal. Extending this to complex and more realistic modulating signals is difficult. We can extend this to look at an example of single tone modulation for some useful information.

An FM broadcast station has a maximum frequency deviation (determined by the maximum amplitude of the modulation signal) of Δf = 80 kHz. The highest modulation frequency f_m is 15 kHz. This yields a modulation index of m = 5.33 and the resulting signal has eight significant sideband pairs. Thus the required bandwidth can be calculated as 190 kHz. For modulation frequencies below 15 kHz (with the same amplitude), the modulation index increases above 5 and the bandwidth eventually approaches $2\Delta f$ kHz = 160 for very low modulation frequencies.

Therefore, we can calculate the required transmission bandwidth using the highest modulation frequency and the maximum frequency deviation Δf_{pk}.

FM Measurements with a Spectrum Analyzer

The spectrum analyzer is a very useful tool for measuring Δf and m and for making fast and accurate adjustments of FM transmitters. It is also frequently used for calibrating frequency deviation meters.

A signal generator or transmitter is adjusted to a precise frequency deviation with the aid of a spectrum analyzer using one of the carrier zeros and selecting the appropriate modulating frequency. In Figure 52, a modulation frequency of 1 kHz and a modulation index of 2.405 (first carrier null) necessitate a carrier peak frequency deviation of exactly 2.405 kHz. Since we can accurately set the modulation frequency using the spectrum analyzer or, if need be, a frequency counter and since the modulation index is also known accurately, the frequency deviation thus generated will be equally accurate.

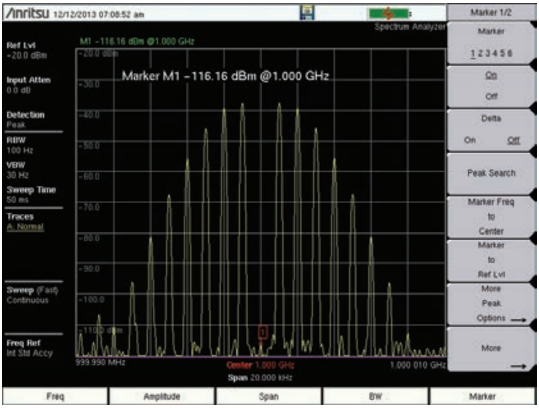


Figure 52

Table 4 gives the modulation frequencies and common values of deviation for the various orders of carrier zeros.

Order of Carrier Zero	Mod Index	Commonly Used Values of FM Peak Deviation								
		7.5 kHz	10 kHz	15 kHz	25 kHz	30 kHz	50 kHz	75 kHz	100 kHz	150 kHz
1	2.405	3.12	4.16	6.25	10.42	12.50	20.83	31.25	41.67	62.50
2	5.52	1.36	1.18	2.72	4.53	5.43	9.08	13.59	18.12	27.17
3	8.65	0.87	1.16	1.73	2.89	3.47	5.78	8.67	11.56	17.34
4	11.79	0.66	0.85	1.27	2.12	2.54	4.24	6.36	8.48	12.72
5	14.93	0.50	0.67	1.00	1.67	2.01	3.35	5.02	6.70	10.05
6	18.07	0.42	0.55	0.83	1.88	1.66	2.77	4.15	5.53	8.30

Table 4

The spectrum analyzer can also be used to monitor FM transmitters (for example, broadcast or communications stations) for occupied bandwidth. Here the statistical nature of the modulation must be considered. The signal must be observed long enough to make capturing peak frequency deviation probable. The MAXHOLD capability, available on spectrum analyzers with digitized traces, is then used to acquire the signal. To better keep track of what is happening, you can often take advantage of the fact that most analyzers of this type have two or more trace memories.

Select the MAX HOLD mode for one trace while the other trace is live. See Figure 53.

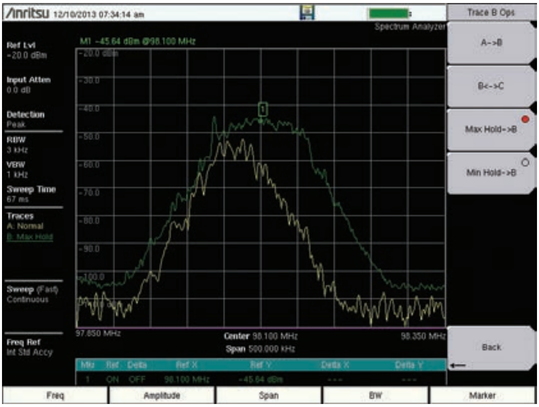


Figure 53

As with AM, it is possible to recover the modulating signal. The analyzer is used as a manually tuned receiver (zero span) with a wide IF bandwidth. However, in contrast to AM, the signal is not tuned into the passband center but to one slope of the filter curve as illustrated in Figure 54. Here the frequency variations of the FM signal are converted into amplitude variation (FM to AM conversion). This method is called slope detection and is not widely used on modern spectrum analyzers since many of them have dedicated FM demodulators.

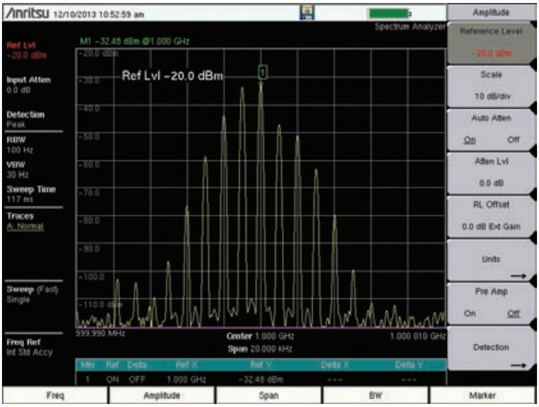


Figure 54

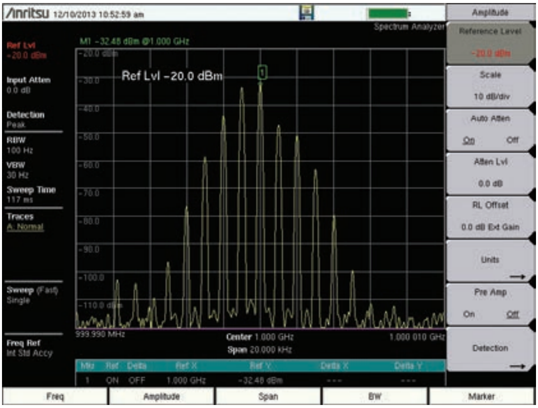
The resultant AM signal is then detected with the envelope detector. The detector output is displayed in the time domain and is also available at the video output for application to headphones or a speaker.

A disadvantage of this method is that the detector also responds to amplitude variations of the signal. The majority of Anritsu spectrum analyzers can provide FM and AM demodulators. In addition, Anritsu handheld spectrum analyzers include SSB signal demodulation with a beat frequency oscillator (BFO) to reinsert the suppressed carrier.

AM Plus FM (Incidental FM)

Although AM and FM are different methods of modulation, they have one property in common; they always produce a symmetrical sideband spectrum.

Figure 55 illustrates a modulated carrier with asymmetrical sidebands. One way this could occur is if both AM and FM or AM and phase modulation exist simultaneously at the same modulating frequency. This indicates that the phase relationship between carrier and sidebands are different for the AM and the angular modulation. Since the sideband components of both modulation types add together vectorally, the resultant amplitude of one sideband may be reduced while the amplitude of the other would be increased accordingly. The spectrum analyzer does not retain any phase information and so in each case displays the absolute magnitude of the result.



Anritsu 12/10/2013 10:52:59 am
Spectrum Analyzer
Ref Lvl
-20.0 dBm
Input Atten
0.0 dB
Detection
Peak
RBW
100 Hz
VBW
30 Hz
Sweep Time
117 ms
Traces
A: Normal
Sweep (Fast)
Single
Freq Ref
Int Std Accy
M1 -32.48 dBm @1.000 GHz
Ref Lvl -20.0 dBm
999.990 MHz
Center 1.000 GHz
Span 20.000 kHz
1.000 010 GHz
Amplitude
Reference Level
-20.0 dBm
Scale
10 dB/div
Auto Atten
On Off
Atten Lvl
0.0 dB
RL Offset
0.0 dB Ext Gain
Units
Pre Amp
On Off
Detection
Freq
Amplitude
Span
BW
Marker

Figure 55

This section explains how to use the Anritsu MS2830A Signal Analyzer to evaluate and analyze various transitional RF signals.

Radio signals (Transient phenomena at Tx start)

This is an example of measuring transient phenomena of radio transmitters.

The antenna output of a small radio transmitter using the 460 MHz band via a dummy load or attenuator is connected to the signal analyzer, and transient phenomena in the emitted radio signal are measured immediately before and after pressing the press-to-talk switch.

Actually, the signal is captured for 20 ms from immediately before the transmission start to observe the change in the spectrum immediately after transmission starts. (Figures 56 to 59 show the transition for a frequency span of 1 MHz and RBW of 10 kHz.)

The sub-trace (lower half) in each figure shows the Power vs. Time for 20 ms; the red and blue time bands in the sub-trace show the spectrum part displayed as the main trace (top half). Although the spectrum rises at the same time as the power rise, in the case of this radio transmitter, the final spectrum distribution has a different shape as the power rises.

Figure 56: Spectrum transition immediately after Start-1

Figure 57: Spectrum transition immediately after Start-2

Figure 58: Spectrum transition immediately after Start-3

Figure 59: Spectrum transition immediately after Start-4

Figure 60 shows the details of the change in power with time with the signal analysis range set to the rise part. It confirms a gentle rise but with a step-shaped form.

Figure 60: Change in power with time (Signal Analyzer)

The same type of results are obtained at Power vs. Time measurement using a spectrum analyzer Zero span (fixed frequency and time on horizontal axis). In this case, since the power is measured within the measured frequency, this method can be used to measure the time until a PLL VCO (voltage controlled oscillator) is locked to the relevant frequency.

Incidentally, Figure 61 shows the spectrum analyzer Zero span measurement results. (Note the different horizontal axis scale.)

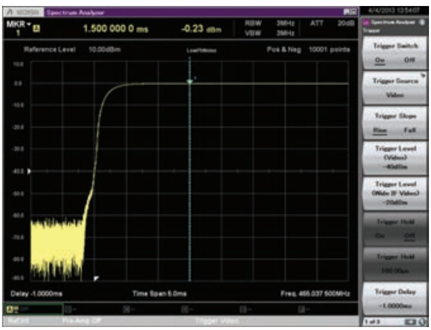


Figure 61: Spectrum analyzer measurement results (Zero span)

Figure 62 shows the same signal, but in this case, it is viewed as a Frequency vs. Time trace.

The small triangle symbol in the figure indicates the signal capture trigger point; in this example the trigger is at the signal rise point so there is a signal to the right of the triangle symbol but no signal to the left.

From this trace, we can see that the frequency immediately after the rise has some slight instability but then stabilizes.

Figure 62: Frequency vs. Time trace

However, although the part on the left before the rise seems to have a huge frequency variation, this part is a meaningless measurement result, because the frequency is uncertain since there is no signal here.

Chirp signals

This is an example of measurement of a chirp signal used by radar, etc.

This type of signal is switched on and off repeatedly while the frequency is changed in a short time period.

Figure 63 shows a split screen with the main trace displaying the Frequency vs. Time and the sub-trace displaying Power vs. Time.

In the Frequency vs. Time display, the analysis time length when the signal is on is set to 2 μs.

Figure 63: Split screen display of chirp signals (Main trace: Frequency vs. Time, Sub-trace: Power vs. Time)

From the Frequency vs. Time trace, we can see that the frequency change is linear over a short time period.

In a Frequency vs. Time trace, the frequency at each time is represented by one point. Since all these points are plotted as one line as shown in the figure, this method is useful for measuring the frequency of a signal where the spectrum changes linearly as shown in this example.

The same state can also be seen on a spectrogram trace, which also displays the simultaneous presence of other signals and spectrum spread.

2-FSK signals

This is an example of measurement of a digital frequency modulated RF signal.

2-Frequency Shift Keying (2-FSK) is a popular modulation method used by many applications, such as automotive keyless entry (RKE), tire pressure monitoring systems (TPMS), etc. In this method, digital 1 and 0 are allocated to two frequencies (f1 and f2) that the signal shifts between.

Until now, 2-FSK signals have been commonly measured using the Max Hold function of a spectrum analyzer (plots maximum value of each frequency at repeated measurement on trace). However, since the spectrum analyzer Max Hold superimposes the repeated measurements, the spectrum lacks data on transient changes and the repeated measurements also take a long time.

In contrast, the signal analyzer requires capture of only one signal to display a spectrum indicating the instantaneous frequency shifts.

The following shows some figures for comparison.

Figures 64 to 67 are the measurement results using a signal analyzer. After capturing the signal for 50 ms, the instantaneous spectrum is displayed for analysis time lengths of 200 μs while changing the start position.

The spectrum shows the two frequency switching states and the spectrum spread during the switch.

Figure 64: 2-FSK signal instantaneous Spectrum-1

Figure 65: 2-FSK signal instantaneous Spectrum-2

Figure 66: 2-FSK signal instantaneous Spectrum-3

Figure 67: 2-FSK signal instantaneous Spectrum-4

Similarly, Figure 68 shows the spectrum when the analysis time is widened to 50 ms.

Since the frequency switches several times during this period, the superimposed spectrum can be viewed.

Figure 68: 2-FSK signal spectrum (Signal analyzer)

On the other hand, Figure 69 shows the measurement results for the same signal using the Max Hold function of a spectrum analyzer.

The measurement requires about 6 s to perform the 100 sweeps each with a sweep time of 60 ms but even with 100 repetitions the entire spectrum cannot be confirmed. Confirming the same spectrum with a spectrum analyzer requires many more sweep repetitions and a lot more time.

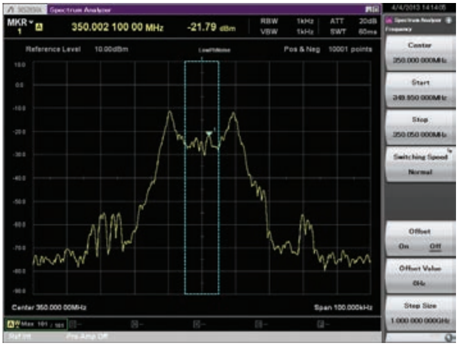


Figure 69 : 2-FSK signal spectrum (Spectrum analyzer)

Frequency hopping

This is an example of Bluetooth signal measurement.

Bluetooth is a short-range wireless communications standard used by small communications devices, such as smartphones.

It uses the Frequency Hopping Spectrum Spread (FHSS) technology in which the frequency changes randomly during a short time period.

Figure 70 shows the spectrogram trace for a *Bluetooth* signal of 100 ms duration.

Figure 70

Since measurement of the 2.4 GHz band is performed with a resolution bandwidth of 31.25 MHz, only part of the *Bluetooth* signal (about 80 MHz bandwidth) is displayed, but it is easy to confirm at a glance how the *Bluetooth* signal frequency changes successively.

A spectrogram is the best trace to use when wanting to intuitively understand a signal with changing spectrum as a whole.

The *Bluetooth*® mark and logos are owned by Bluetooth SIG, Inc. and are used by Anritsu under license.

Noise

This is an example of overall noise analysis using the various signal analyzer/spectrum analyzer functions.

The measurement target is electromagnetic noise generated by a LED light bulb. An EMI probe (small loop antenna in Figure 71) is placed near the LED light to detect surrounding EMI.

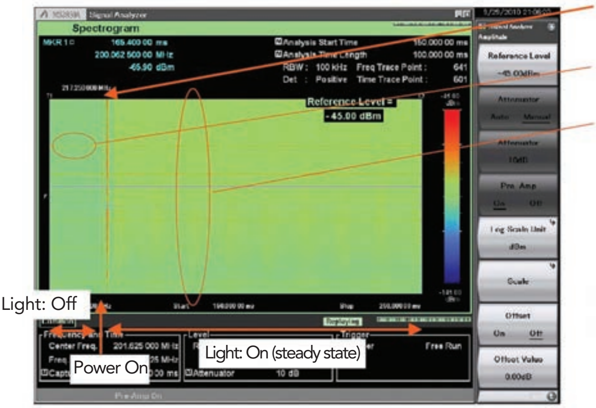
Figure 71: Anritsu EMI Probe (MA2601B)

First, a wide frequency band is swept from 9 kHz to 1 GHz using the spectrum analyzer function to detect the noise frequency band (Figure 72). This is because noise is distributed across a wide frequency range generally.

Figure 72 : Sweep measurement using spectrum analyzer

The yellow trace in the above figure is when the LED light is off. The spectrum seen at this time is coming from some external source, such as a TV broadcast wave. On the other hand, the blue trace in the figure is the spectrum when the LED is on. Noise components can be seen centered around 100 MHz to 300 MHz. Consequently, it is best to set this range as the signal analyzer analysis range.

Figure 73 shows the entire spectrum trace when the signal analyzer frequency range is set to 186 MHz to 217.25 MHz (analysis bandwidth: 31.5 MHz) with an RBW of 100 kHz and the noise signal is captured for 250 ms including the period before the LED light is lit.



- Frequency range: 186 MHz to 217.25 MHz
- LED light: Off to On (steady state)

Figure 73: Noise spectogram trace

The noise has a wide spectrum and is seen as wide traces in the spectrogram.

Based on this spectrogram, there is large wideband noise generation at power-on that stops, but then wideband noise also reoccurs when the LED light is lit.

Figures 74 and 75 compare the instantaneous spectrums of the captured signals for 500 μs at power-off and at power-on, respectively, for the specified frequency. Compared to power-off, the noise at power-on clearly rises by close to 20 dB.

Figure 74: Power-off spectrum

Figure 75: Instantaneous spectrum at power-on

In addition, Figure 76 shows the Power vs. Time trace when the LED is lit, confirming that the noise is pulsing periodically.

Figure 76 : Power vs. Time trace when LED lit

Additionally, for confirmation, Figure 77 shows the Amplitude Probability Distribution (APD) of the noise as a CCDF/APD trace at the power-on instant. It follows a gentle curve, suggesting a single noise source. If there were multiple noise sources, the curve should be changing.

Figure 77 : Noise APD

Since the measurements so far indicate the wideband noise is periodic, the signal is captured for a 100-ms period at a center frequency setting of 185 MHz, span of 10 MHz, and RBW of 30 kHz to analyze the noise while the LED is lit (Figure 78).

Figure 78 : Noise when LED lit

Figure 78 : Noise when LED lit

Pulse Measurements

A spectrum analyzer is an important instrument for measurements on pulsed or bursted signals. Many of the different parameters of such sig-nals can be directly measured with a Spectrum Analyzer. This tutorial ex-plains the use of the Anritsu Signal Analyzer family (MS269xA / MS2830A) and the handheld MS2720T for the measurement and characterization of such signals.

Motivation

Communications systems for information transfer, which for a long time have mainly been implemented in analog form, are increasingly being replaced by digital components and systems. The latter often use pulse-modulated signals, for instance in television, radar and mobile radio. Due to the spectral distribution of such signals, a spectrum analyzer used to measure the signals has to fulfill special requirements.

Theoretically, the energy of pulse-modulated signals is distributed over the whole spectrum. The measured energy strongly depends on the resolution bandwidth and on the point of measurement in the spectrum. If the SI-Function is measured close to a null in the envelope, overdriving of the input stage may be caused as a result of incorrect setting. The total energy spectrum is applied to the input stage if no preselection filters are used. This reduces the spectrum and applies the spectrum to the mixer of the first conversion stage in 'slices'. Modern spectrum analyzers feature low nonlinearities and high overload capacity (high intercept points of 2nd and 3rd order and high 1 dB compression). Moreover, they are equipped with internal overload detectors used for automatic correction of the analyzer settings in order to optimize the dynamic range and shift it into a non-critical level range with the aid of automatic RF attenuation settings (auto range function). This ensures ease-of-operation for the user and reliable measurement.

Pulsed signals are difficult to characterize, because often the pulse width and pulse repetition frequency are not constant which effectively eliminates the simple RF power meter as a tool for calculating pulsed-signal peak power from mean power. Often many parameters must be measured in order to effectively characterize a pulsed signal, e.g.:

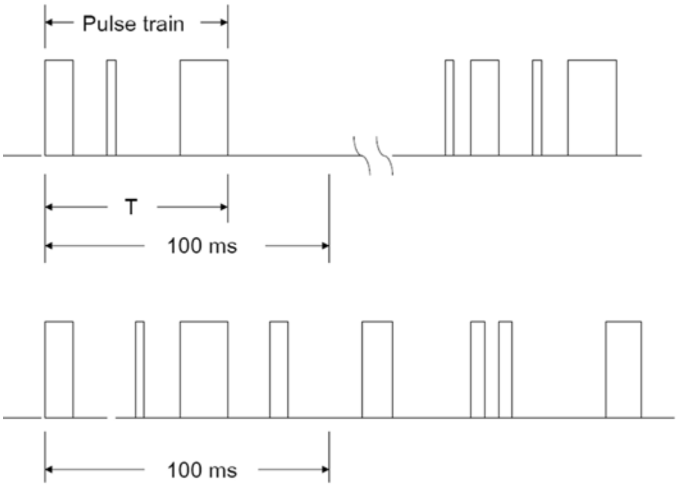


Figure 79

- Peak and average power,
- Pulse shape,
- Pulse profile
  rise time,
  fall time,
- Pulse width,
- Pulse off time,
- Pulse repetition interval (PRI),
- Pulse repetition frequency (PRF),
- Occupied spectrum
- Duty cycle

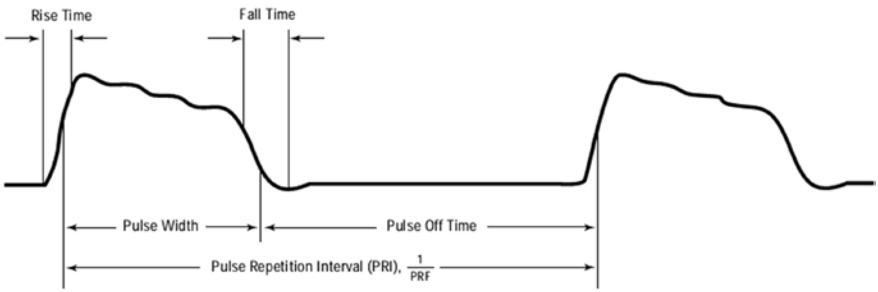


Figure 80: Pulse signal name conventions

Pulse Fundamentals

We will start our discussion by reviewing the formation of a square wave from a fundamental sine wave and its odd harmonics. The description of pulse signals is based on an ideal, periodic rectangular pulse sequence. You will probably recall plotting a sine wave and its odd harmonics on a sheet of graph paper, then adding up all of the instantaneous values. If enough harmonics were plotted at their correct amplitudes and phases, the resulting waveform begins to approach a square wave.

A rectangular pulse is merely an extension of this principle where the relative amplitude and phase of the harmonics, both odd and even, are changed. In this manner we can plot an infinite number of waveforms. The spectrum analyzer effectively unplots complex waveforms and presents the fundamental frequency and each harmonic contained in the waveform.

The Fourier transform of a rectangular pulse has the familiar (sin x)/x shape as defined by:

$$V(f) = \tau \frac{\text{Sin}\,[\omega\,(\tau/2)]}{\omega\,(\tau/2)}$$

The (sin x/x) nulls occur at multiples of 1/τ. In the case of the single pulse, the spectrum is a continuous spectrum with no individual frequency lines. The swept tuned spectrum analyzer is not capable of viewing these single event transients; however, an FFT analyzer or vector signal analyzer (VSA) can be used, provided it has sufficient bandwidth (at least BW ≧ 1/τ).

A pulse train is produced by repeating the pulse at regular time intervals. Since this waveform is periodic, it can be expanded into a Fourier series to determine its harmonic content, as follows:

$$V(f) = \frac{\tau}{T} + \frac{2\tau}{T} \sum_{n=0}^{\infty} \frac{\text{Sin}\,[\omega\,(n\pi\tau/T)]}{n\pi\tau/T)}$$

As displayed in Figure 81 the average value of the waveform, or DC component, is simply τ/T. The harmonics fall at multiples of the waveform frequency, or PRF, which is 1/T. Again the waveform takes on the familiar (sin x)/x characteristic. Just as with the single pulse the spectral envelope has nulls that occur at integer multiples of $1/\tau$. The amplitude of the spectrum is proportional to the pulse width (τ), because there is greater power in the waveform as pulse width is increased. To be more precise, the amplitude of the spectrum depends on the duty cycle of the waveform, which is the ratio of pulse width to period (τ/T).

Up to this point we have only talked about video pulses. Now we will use the video pulse to amplitude modulate an RF carrier. As with single-tone AM, modulation side-bands are formed symmetrically above and below the carrier frequency. Since the spectrum of the modulating pulse is made up of many individual spectral lines, or tones, the resulting modulated RF signal will be composed of many sidebands which are typically referred to as spectral lines. Essentially, the modulation process simply translates the video pulse's spectrum up to the carrier frequency.

As shown in Figure 82, the individual spectral lines represent the modulation product of the carrier and the spectral lines of the modulating pulse, where the line spacing is equal to the PRF of the modulating pulse and the sin(x)/x nulls fall at integer multiples of $1/\tau$. As the PRF is increased the spectral line density increases and as the pulse width is reduced the sin(x)/x lobes broaden. These relationships are shown graphically again in Figure 82.

Notice in the graphics used thus far that the spectral lines extend both above and below the baseline. This is caused by the harmonics of the modulating signal having 180° phase relationship to the fundamental of the modulating waveform.

Since the spectrum analyzer uses simple envelope detection these phase relation-ships are lost and the analyzer will invert the negative going lines displaying all lines above the baseline. However, the vector signal analyzer (VSA) can display the spectral lines in their proper orientation.

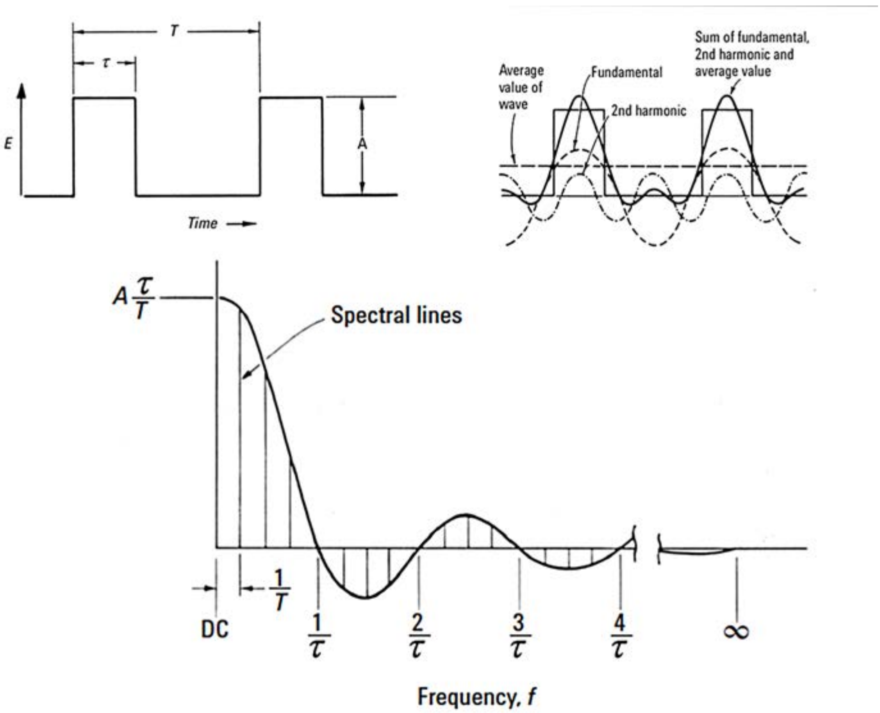


Figure 81: Line and envelope spectra of a rectangular video pulse train shown in time domain and in frequency domain. The envelope of the spectral line is an SI-Function and decaying proportional to 1/f.

The smallest frequency f is the fundamental, corresponding to the reciprocal value of the period T, also called Pulse Repetition Frequency (PRF).

E 3: $$f = \frac{1}{T}$$

The first null of the SI-Function occurs at the reciprocal value of the pulse duration:

E 4: $$f_{si1} = \frac{1}{\tau}$$

Further nulls follow at $f_n = n \quad f_{si}$ intervals.

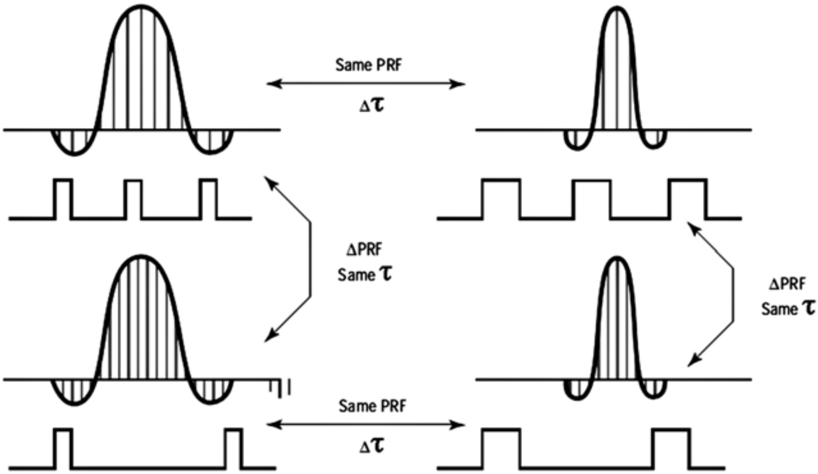


Figure 82: Changing PRF or PW changes frequency spectrum (envelope and spectral lines)

To summarize and without mathematics we can state:

- The frequency range between any spectral line is equal to PRF (inverse of the PRI)
- The nulls are equal to the inverse of the pulse width τ
- If the PRF is decreased, the number of spectral lines increases

With regard to the Pulse Width (PW) we can state:

- The wider the pulse width, the narrower the frequency-domain peak will be
- The smaller the pulse width, the larger the bandwidth will be
- If signal's PW is increased, the frequency spectrum shrinks
- Increasing the pulse width while decreasing the PRF produces a smaller spectrum with more spectral lines

While the Fourier representation furnishes contributions from $-\infty$ to $+\infty$ and the coefficients may also have a negative sign, the spectrum analyzer only represents positive frequencies according to their magnitude. Thus the Fourier Analysis is resulting in a SI-Function around modulated carrier frequency f_0 (Figure 83).

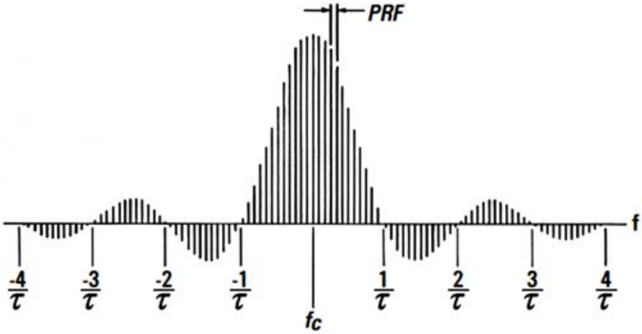


Figure 83: General spectral display (SI-Function) after Fourier analysis with modulated carrier frequency f0

Depending on the measurement or resolution bandwidth (RBW), the following three cases are possible when using a frequency-selective Spectrum / Signal Analyzer for the pulse spectrum measurement:

- Resulting Line Spectrum,
- Resulting Pulse Spectrum with a shape of an SI-Function
- Envelope Spectrum without SI-Function shape

In the following we will investigate the differences between these three different cases.

Line Spectrum

The Line Spectrum is realized when the Spectrum Analyzer RBW is narrow compared to the frequency spacing of the input signal components. When this conditionis satisfied all of the individual frequency components can be resolved, because only on efrequency line is within the SPA RBW at a time.

In this case the SPA display is a frequency domain display of the actual Fourier components of the input signal whereby each signal component behaves as a CW signal.

Rule of thumb for line spectrum

- RBW < PRF (1/T)
  - RBW < 0.1 - 0.3 PRF
  - Line spacing is independent from Sweep Time
  - Line amplitude will not change as long as RBW << PRF
  - In the case of the line spectrum the number of lines does not vary as a function of

the bandwidth or frequency span, the amplitude remains constant
- Valid for IF Filter Skirt Factor (K = 4 to 4.5:1) of modern digital-IF based spectrum analyzers

The measurement of nulls in the pulse spectra in practice delivers not always distinct results, because they are somewhat blurred. The reason lies in the asymmetries of real signals that cannot be avoided, since in contrast to the theoretical ideal rectangular pulses, the finite exponential rise and fall times of the real pulses have to be taken into account. For our example we will deal from now on with some slightly different pulse widths, dependent on how they have been measured.

Here in our example with τ = 380 ns and PRF = 1 kHz the used RBW must be smaller than 1 kHz. Using the above rule of thumb it is useful to operate between 100 and 300 Hz RBW.

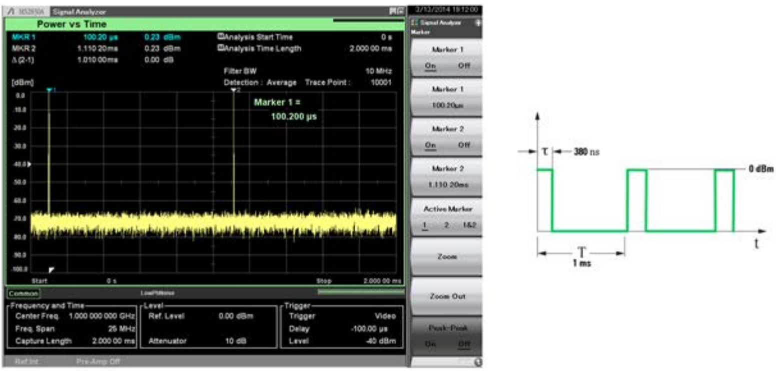


Figure 84: Measurement of 1 kHz Pulse Repetition Frequency in Power versus Time operation mode

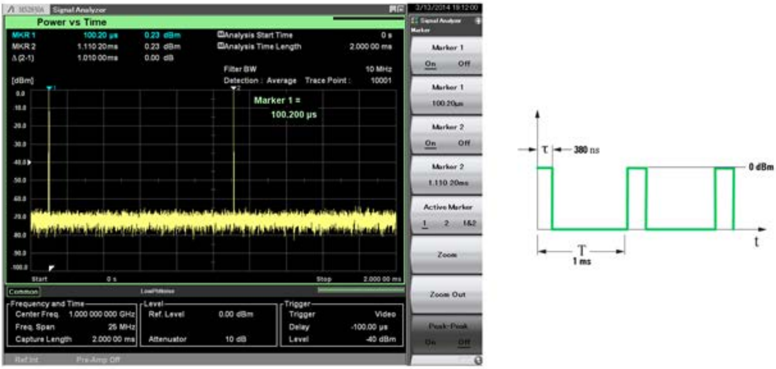


Figure 85: Measurement of 380 ns Pulse Duration (τ)
Power versus Time operation mode

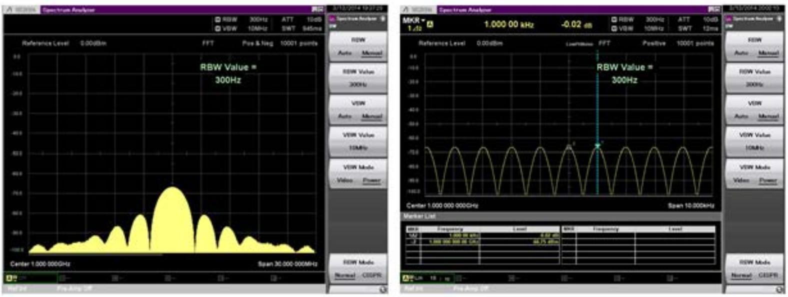


Figure 86: Left - Entire pulse within a 30 MHz Span and 300 Hz RBW resulting in a SI-Function shape

Right - Span reduced down to 10 kHz with RBW 300 Hz - Line Spectrum with equidistant spectral lines representing the PRF is visible

Pulse Spectrum

The bandwidth RBW is greater than the spacing Δf of the spectral lines, but smaller than the spacing 1/τ of the first null of the envelope SI-Function from the carrier fre-quency. The spectral lines cannot be resolved and the amplitude height of the enve-lope depends on the resolution bandwidth. This makes sense as the amplitude de-pends on the number of spectral lines collected within the measurement bandwidth.

Rule of thumb for Envelope Spectrum

- 1.7 PRF < RBW < 0.2/τ
  - Spectral lines are not resolved and the amplitude height of the envelope depends upon the RBW
  - In the case of the pulse spectrum the number of lines varies as a func-tion of the RBW and not as function of the frequency offset. The displayed amplitude increases with the RBW due to the larger energy component within the measurement bandwidth.
  - The amplitude will change linearly as the RBW is increased; 6 dB am-plitude rise for each doubled RBW or in our example 20 dB for a tenfold RBW.

Here in our example with PRF = 1 kHz and τ = 425 ns (380 ns) the used RBW must be in the range of 1.7 kHz till 470 kHz. In Figure 87 you can see an example for pulse form spectrum with RBW = 10 kHz. On the right side you can see afore discussed dependency of RBW and power level (Yellow trace 10 kHz, blue trace 100 kHz RBW).

In practice it is useful to operate pulse spectrum because it is either impossible or not desirable to perform high resolution line-by-line analysis. A good example for this is the narrow pulse width and low PRF pulse train of real RADAR signals. Additionally you get a signal-to-noise ratio advantage operating pulse spectrum. Since signal amplitude increases linearly as we increase RBW we achieve 6 dB signal increase each time we double the RBW. Whereas the noise level only increases by 3 dB (not valid for CW or line spectrum analysis!).

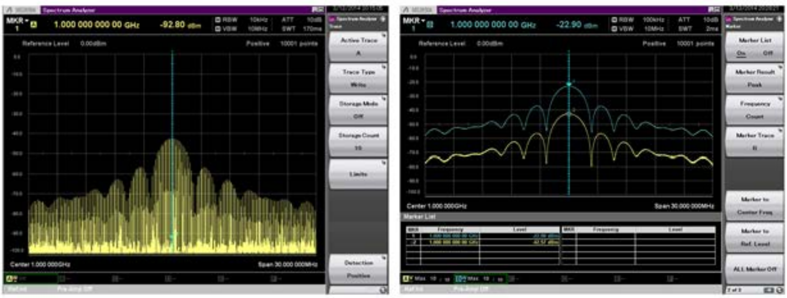


Figure 87: Pulse Spectrum for RBW of 10 kHz (left), 20 dB spectrum growth for 10 fold RBW (right)

Envelope Spectrum

As mentioned previously, when analyzing the pulse spectrum we use RBW greater than the PRF, but there is a limit. As the RBW is increased, we will start to notice that the nulls of the SI-Function envelope will become less defined and will total disappear. In our example you will see this starting above 300 kHz RBW and becomes significant from 1 MHz onwards. In Figure 88 you will clearly recognize that the SI-Function disappeared completely. The good thing is that in this case the analyzer will measure the true peak power of the pulse, approximately 0 dBm.

Rule of thumb for Envelope Spectrum

- RBW > $1/\tau$
  - For this case where the measurement bandwidth is greater than the null spacing's on the signal spectrum envelope, the amplitude distribution of the signal cannot be recognized, because the entire spectrum falls within the bandwidth
  - With a further increase in bandwidth, the response approaches the time domain function of the pulse

Here in our example (with τ = 425) the PRF must be higher than 2.35 MHz.

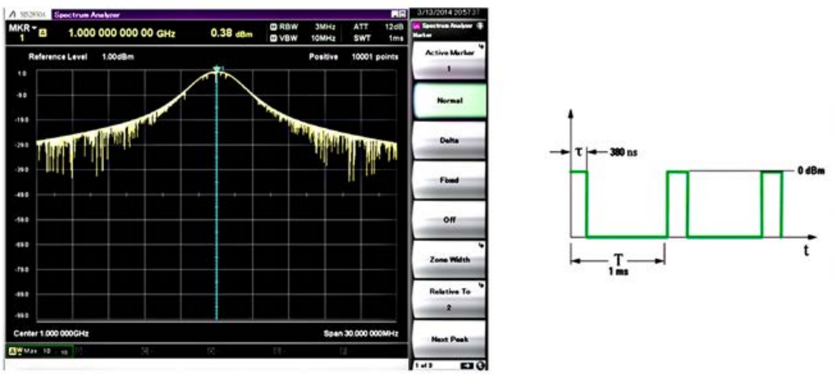


Figure 88: Left: As RBW is increased spectral nulls are blurred and disappear finally (RBW 3 MHz with Maxhold)

Right: In frequency domain the pulse looks more and more as in the time domain (RBW 20 MHz)

Brief summary

The below Table 1 summarizes the characteristics of the Line and Pulse Spectrum. One important hint is always to consider the total power that such a pulse generates. Be cautious – in no case should the peak input power to the SPA mixer exceed the specified 1 dB compression point.

	Line Spectrum	Pulse Spectrum
RBW	RBW < 0.3 PRF	1.7 PRF < RBW < 0.2τ
Sweep Time	TSweep > SPAN/(RBW)^2	TSweep > PRT
Desensitization	20 log10(τ/T)	20 log10(τ K RBW)
Number of lines/div	Changes with SPAN not with T_{Sweep}	Changes with T_{Sweep} not with SPAN

Table 5: Line and Pulse Spectrum Summary

Pulse desensitization – Peak Power Measurements

Sometimes the issue of pulse desensitization is referred to in terms of pulse spec-trum analysis. The issue is that when the modulation is applied to the carrier, the peak level of the envelope is reduced, appearing that the signal has been reduced in overall power.

The apparent reduction in peak amplitude occurs because adding the pulse to the signal and modulating it with a square wave results in the power being distributed between the carrier and the sidebands. As the level of the modulation increases, so does the level of the sidebands. As there is only limited power available and each of the spectral components, i.e. carrier and sidebands, then contains only a fraction of the total power.

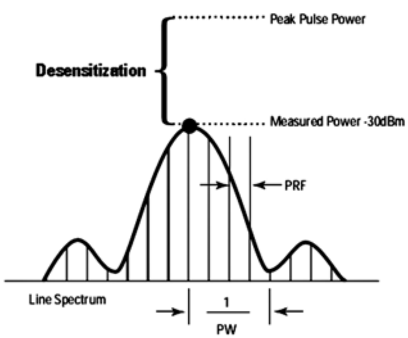


Figure 89: Example for Pulse

The overall effect as seen on a spectrum analyzer is that the peak power reduces, but it is spread over a wider bandwidth. This is because a pulse, by definition, is not on all of the time, while a spectrum analyzer is averaging the signal over time, which reduces the displayed magnitude of a pulsed signal (a pulsed signal is not on at all times, the energy will not completely "fill" the spectrum on a single sweep).

It is possible to define a pulse desensitization factor α. With this desensitization factor we can determine the peak power of the measured signal, otherwise we would get always a wrong result. This can be described in the following equation:

E5 $\alpha_L[\text{dB}] = 20\log_{10}\left(\frac{\tau}{T}\right) = 20\log_{10}(\tau\ \text{PRF})$ For a line spectrum

It should be noted that this relationship is only really valid for a true Fourier line spectrum. For this to be applicable the RBW of the analyzer should be < 0.3 PRF. The average power of the signal is also dependent on the duty cycle as the power can only be radiated when the signal is in what may be loosely termed the "ON" condition. This can be defined by the Duty Cycle equation below:

E6
$$\frac{P_{avg}}{P_{Peak}} = 10\log_{10}(\tau \bullet \text{PRF})$$

$$P_{avg} = P_{Peak} + 10\log_{10}(\tau \bullet \text{PRF})$$

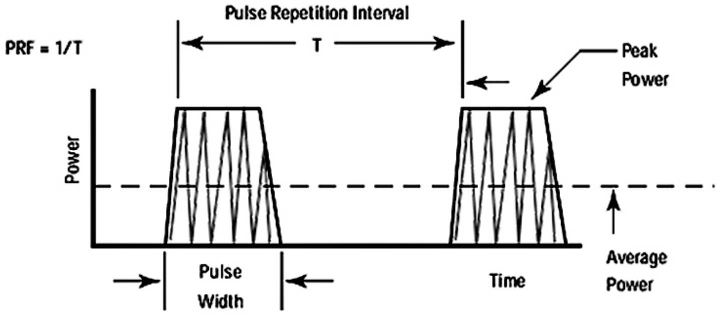


Figure 90: Pulse definitions

Where: α= Pulse "desensitization factor"
T= pulse repetition rate (PRR)
PRF = Pulse Repetition Frequency (1 / T)
τ = effective pulse width taking account of rise and fall times
P_{Avg} = Average power over a pulse cycle
P_{Peak} = Peak power
K = Filter skirt factor

Unlike the line spectrum, the pulse spectrum envelope amplitude changes as the RBW is changed. This makes it difficult to make accurate amplitude level measurements. By applying again a pulse spectrum desensitization factor αP it is possible to determine the actual amplitude.

E7 $\alpha\, P[dB] = 20\log_{10} (\tau \bullet K \bullet RBW)$ For a pulse specturm

The Filter Shape Factor K depends on the type of the IF filter. Typical examples are K = 1 for Gaussian filters and K = 1.5 for rectangular filters. For pulse signal measurements, a compromise has to be found since with small resolution bandwidths the displayed amplitude may become too small, whereas with large resolution bandwidth the displayed amplitude will be larger but the resolution degraded to an increasing extent. In practice, the following value has been empirically determined:

E8 $\tau \bullet RBW = 0.1$

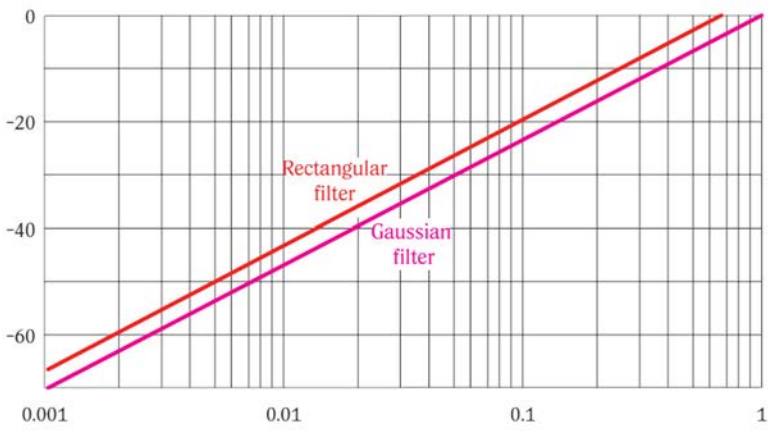


Figure 91: Amplitude loss as a function of time/bandwidth product τ•B

Let's understand the above stuff based on a real example: A pulse of the duration τ = 380 ns and PRF of 1 kHz (= 1/T), corresponding to a period T = 1 ms, is measured with a Gaussian filter (K = 1) of bandwidth RBW = 300 Hz. The "Rule of thumb for line spectrum" applies (RBW < 1/T), so we have a line spectrum. Equation E5 then gives:

E9 α [dB] = 20•log(425 ns/ 1 ms) = -67.43 dB

Accordingly, the displayed amplitude value of the un-modulated carrier would be 67.43 dB higher. As we can read out of Figure 92 the center frequency level is measured with -66.96 dBm + 67.43 dB is resulting in a peak power of 0.47 dBm. This corresponds also very nice with the result out of Figure 88. Let's proof this with some measurements using Time Domain analysis mode.

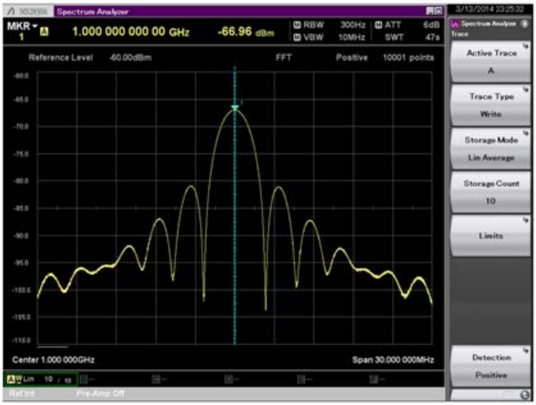


Figure 92: Line Spectrum situation - Center frequency amplitude level -66.96 dBm

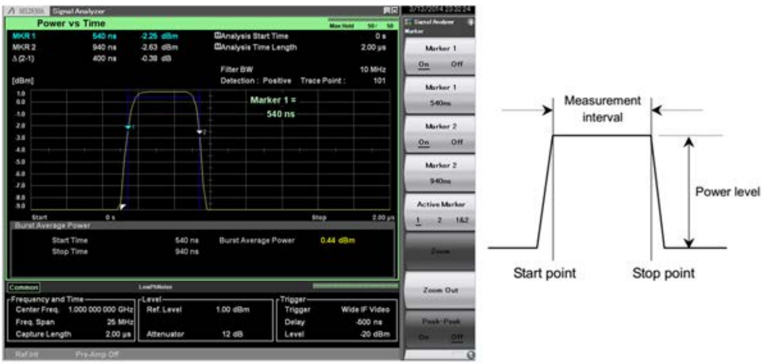


Figure 93: Signal under test measured in Time Domain mode with Burst Average Power feature

In order to proof the result we will use the Burst Average Power feature of the FFT based Spectrum Analyzer (SA operation mode). In this mode the MS2830A can analyze simultaneous input signals by capturing IQ data for a certain time interval and calculate the resulting Power over Time by integrating stepwise the power levels.

As we can see from the results visible in Figure 93 the resulting Burst Average Power is again approximately 0.44 dBm. This correlates very nice with our previous number based estimate.

In Figure 94 it is apparent that we have used an averaging factor of 10. Principally averaging should be avoided when making measurements on pulsed signals. Pulses have a high peak and a low average value (depending on mark-to-space ratio). In order to avoid too low display levels, the video bandwidth should be selected much greater than the resolution bandwidth. Otherwise amplitude errors due to too small VBW will be observed.

VBW = 10 MHz → P = -66,14 dBm
VBW = 10 Hz → P = -62,74 dBm
Delta: → Δ = 3,4 dB

Figure 94: Signal under test measured with RBW 1000 kHz / VBW 10 MHz and 10 Hz and an average factor of 10

Average Power Measurements

Average power can be measured by integrating power over a specified bandwidth. Since most of the signal power is contained in the main lobe and the first side lobes of the SI-Function, the integration bandwidth should be at least ($4/\tau$) wide. Increasing the integration bandwidth will provide only little improvement in the measurement. Assume the energy, E, contained in every pulse is constant. Power is just the time rate of change of the energy flow (energy per unit time).

- Definition of Peak Power:
  Rate of energy flow in every pulse
  $$P_{Peak} = \frac{E}{\Delta t}$$
- Definition of Average Power:
  Rate of energy flow averaged over one full period (1/T)
  $$P_{Avg} = \frac{E}{T}$$

Solving the equations towards E gives us the previously known equation E6. Thus knowing the duty cycle and the peak power gives us finally the average power.

Using the Channel Power measurement feature of the SPA should therefore reveal similar or same results. According to Figure 93 we measured a peak power of 0.44 dBm.

E10 $\quad P_{Avg} = 0.44 \text{ dB} + 10\log_{10}(425 \text{ ns} \bullet 1 \text{ kHz}) = 0.44 - 33.7 = -33.26 \text{ dB}$

We are able to proof the result using the Channel Power feature of the Spectrum Analyzer as given on Figure 95.

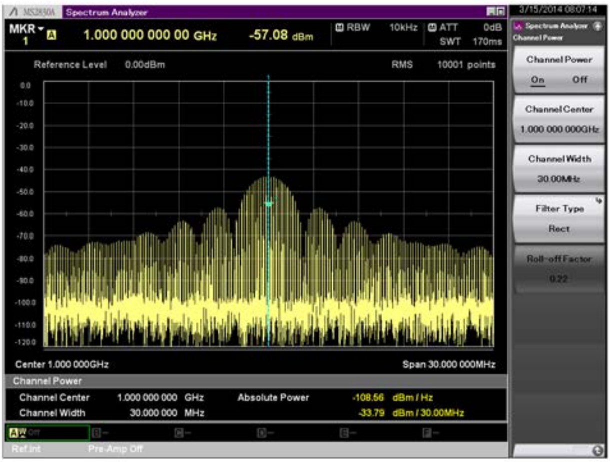


Figure 95: Average Power measurement using Channel Power feature

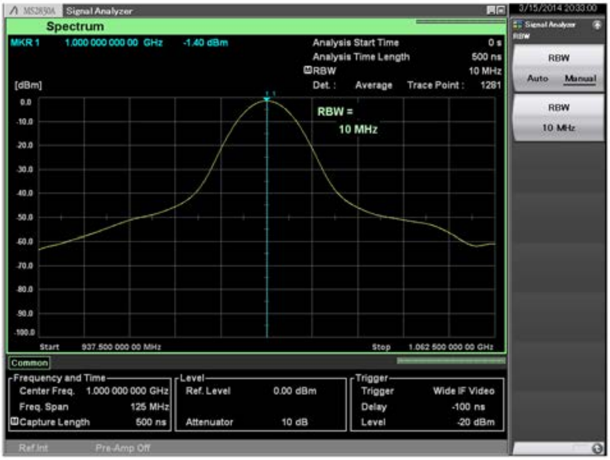


Figure 96: Pulse signal captured in SA operation mode with 500 ns capture length, 10 MHz RBW

Compared to Figure 88 with approximate 30 seconds Maxhold and averaging the above Figure 96 took just a blink and delivers nearly the correct peak power. Think also about the very precise and fast pulse width measurement that you gain with Capture & Replay operation (Picture 6)

Spectrum Master MS2720T Burst Detect Mode

Burst detect is a sweep method that makes it easy to see short-duration, bursty signals, such as those emanating from an improperly installed cell phone booster. It can also show the envelope of a Wi-Fi signal – which is basically anything that occurs infrequently.

A 1% duty cycle is enough to detect a bursty signal. As many as 20,000 measure-ments per second – thousands of times faster than a normal FFT – can be made with the Burst Detect method. The result is that users can see a 200 microsecond pulse every time, making it much easier to find burst signals. Burst detection is accomplished by capturing a portion of the spectrum using digital signal processing techniques and is not a swept measurement; the entire span to be observed is captured simultaneously. The captured span can be as wide as 15 MHz and as narrow as 20 kHz. Within the digital signal processor, a max hold operation is performed on the spectra. About 8 times a second the display is updated with the latest max hold values, which are then reset. The display update rate is high enough so that the hardware max hold is useful and fast enough that the display updates frequently; changes in the signal being observed will be quickly seen.

This automatic operation allows a user to do direction finding on a very low duty cycle signal, something that had not been practical previously. While Burst Detect can be used with several detection modes, for most practical purposes, using the peak detector makes the most sense. The RBW is automatically set when using Burst Detect per the following table. When using Burst Detect, the VBW is always set to 10 MHz. Below a span of 20 kHz Burst Detect isn't used.

Bandwidth	RBW
10 to 15 MHz	100 kHz
3 to 9.9 MHz	30 kHz
1 to 2.9 MHz	10 kHz
300 kHz to 999 kHz	3 kHz
100 kHz to 299 kHz	1 kHz
20 kHz to 99 kHz	300 Hz

Table 6: Burst Detect Analysis Bandwidth versus RBW

After the presence of an interfering signal is detected, you may be able to gain significant insight by looking at the signal in zero span. Doing so lets you see timing information for the signal but doesn't show you the spectrum, so it is generally a step later in the process of identifying a signal.

Here is an example of a signal measured in zero span. To see a stable presentation of the signal video triggering is used with the trigger amplitude set to about 15 dB below the peak of the signal, triggering on the rising slope (which is the default) and the trigger delay set to -1% to place the rising edge of the signal on screen.

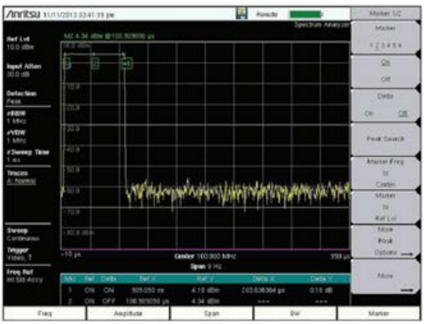


Figure 97: Narrow Signal in Zero SPAN

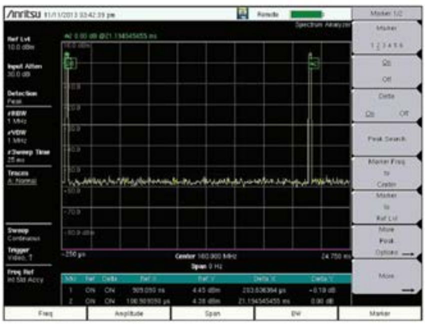


Figure 98: Narrow signal showing low duty cycle

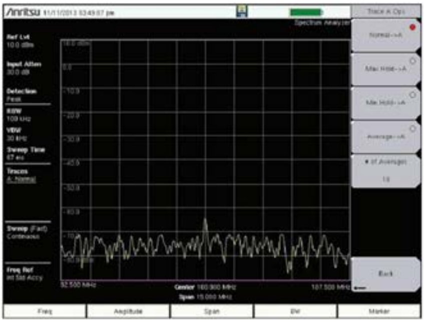


Figure 99: *Low duty cycle signal with Peak Detector*

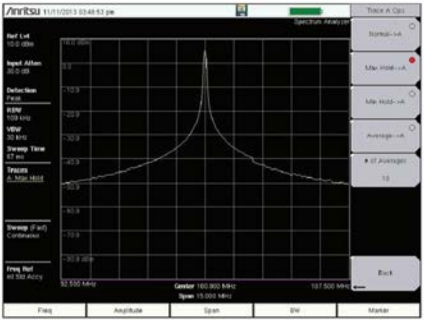


Figure 100: Low duty cycle signal using Burst Detect

The signal, when measured using the peak detector in Fast, Performance, or No FFT sweep mode, often shows nothing as shown in Figure 99 and sometimes will show a portion of the signal. Using one of those detection modes, Performance and No FFT, will start to show portions of the signal quickly, but it takes a long time to fill in the shape of the signal. Fast Sweep mode fills in wide sections of the display at a time, but it also takes quite a while to see the entire signal. Also, Max Hold is not particularly useful when doing direction finding because of the need to clear max hold whenever you want to take a new measurement. This makes interference hunting or other direction finding very tedious.

Measuring the envelope of a Wi-Fi signal is easy and quick with Burst Detect. In the following picture, the green trace is a Wi-Fi signal that has been captured using Max Hold. The yellow trace is a live trace using Burst Detect. The live Burst Detect trace is updated every 67 ms, whereas the Max Hold trace required from about three to about fourteen seconds depending on whether the measurement was done using fast sweep, performance sweep, or No FFT sweep. One minor limitation of Burst Detect is that the maximum specified span is 15 MHz.

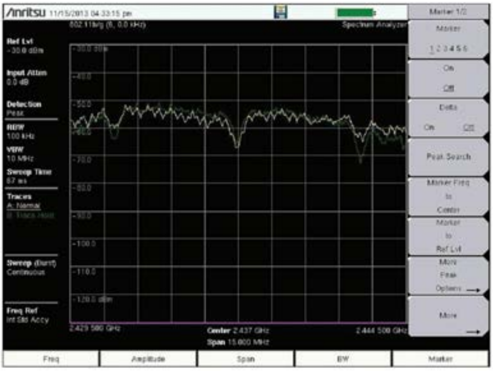


Figure 100: WiFi signal measured with Burst Detect

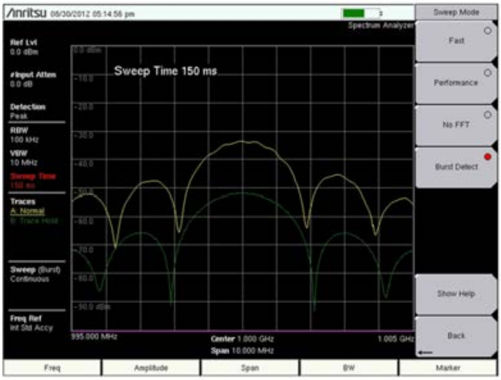


Figure 101: Test signal acc. to page 6 in Burst Detect mode

For anyone desiring to look for signals that are repetitive but occur infrequently, Burst Detect is a very useful feature.

Measurement Examples

The measurements described in this section are generally available 'one button' functions on modern, high performance spectrum analyzers as but may not appear on all the available models.

Intermodulation Distortion

Signals generated by intermodulation distortion appear as signals that are separated from the original signals by the frequency difference of the original signals. The level of this intermodulation distortion depends on the levels and frequencies of the input signals. When two signals are input, the distortion is observed as 3rd order distortion, and when the input signal level is decreased by 10 dB, the distortion decreases by 30 dB. Figure 101 shows this relationship and the point (where the input signal meets the distortion component) is called the intercept point.

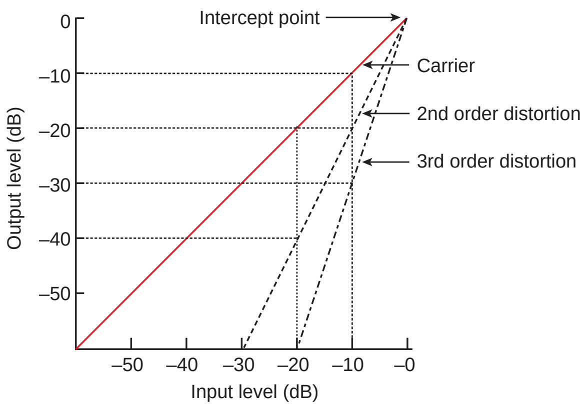


Figure 101

Intermodulation distortion is even generated in the spectrum analyzer itself and this distortion component is determined by the mixer input level. Consequently, when measuring intermodulation distortion using a spectrum analyzer, it is necessary to take care about the mixer input level. It is possible to determine whether or not the DUT or the spectrum analyzer is generating the distortion by observing whether or not the distortion component changes when the spectrum analyzer input attenuation value is varied.

When the spectrum analyzer is generating the distortion, the distortion component changes by 15 dB when the input attenuation is varied by 5 dB. Consequently, in this case, it is necessary to increase the value of the input attenuator to the point where the distortion does not change. In addition, when two signals are input to the DUT, the two signal sources cause mutual interference and hence intermodulation distortion occurs. To distinguish this, confirm whether or not the distortion changes by a factor of 3 relative to the attenuation value when the attenuator in front of the DUT is varied. When the distortion component does not change by a factor of 3, insert an isolator between the signal combiner and the signal sources.

C/N measurement

The output signal from equipment such as a signal generator is not a pure sine wave, and as well as harmonic components, it includes noise of amplitude components and frequency components. These are generally called AM noise and FM (phase) noise. Generally, the AM noise is lesser in magnitude in comparison to the FM noise so measurement of FM noise is explained here.

The FM noise exists just above and below the carrier wave as shown in Figure 102 and is expressed as the ratio of the single sideband phase noise power to the carrier wave power within a 1 Hz bandwidth for a specified frequency offset from the carrier. When a spectrum analyzer is used, the carrier wave power and the sideband noise can be viewed directly on screen. However, the following points must be noted when using a spectrum analyzer.

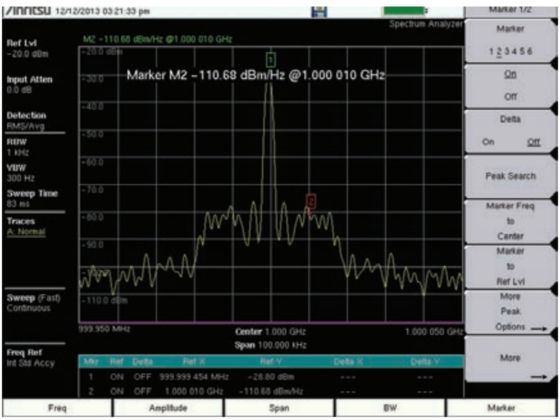


Figure 102

1) Averaging noise power - Since a spectrum analyzer has a peak hold circuit in front of the A/D converter, when noise is measured, the maximum power of the noise over the sampling period is displayed. Generally, noise is evaluated as the average value of the power against time. Consequently, it is necessary to use a sampling detector and to narrow the video bandwidth in order to average the noise power.

2) Conversion for noise bandwidth - Since the value of the measured noise power depends on the noise bandwidth used, correction for a 1 Hz noise bandwidth is required.

3) Correction of average noise value - With a spectrum analyzer, since the signal is logarithmically converted and envelope detected, the average value of the noise appears to be lower than the actual RMS noise value, so this value must also be corrected.

Occupied Frequency Bandwidth

A common measurement carried out on radio transmitters is that of occupied frequency bandwidth (OBW). This measurement calculates the bandwidth containing the specified amount of the total integrated power of the displayed spectrum. However there are two different methods of calculation depending on the technique used to modulate the carrier.

a) XdB Down method

The occupied frequency bandwidth is defined as the bandwidth between the upper and lower frequency points at which the signal level is XdB below the peak carrier value (Figure 103).

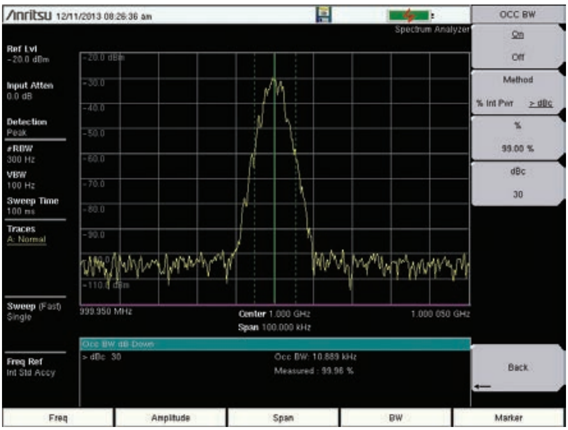


Figure 103

b) N% method

The occupied frequency bandwidth is calculated as the bandwidth containing N% of the power transmitted where N can be between 1% and 99%. A typical example is shown in Figure 104.

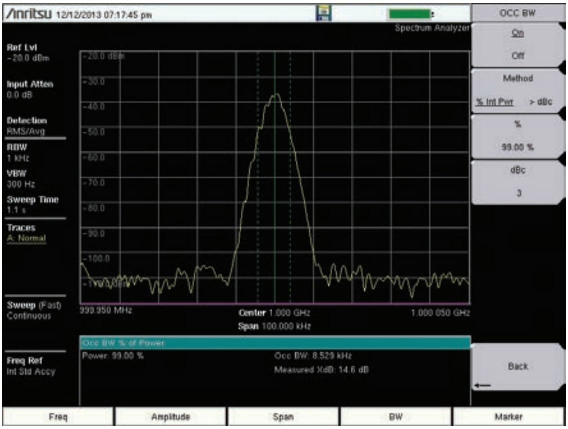


Figure 104

Adjacent Channel Leakage Power

Another common transmitter measurement is that of adjacent channel leakage power. This is defined as the ratio of the amount of leakage power in an adjacent channel to the total transmitted power. In order to calculate the upper and lower adjacent channel values, the spectrum analyzer needs three parameters to be specified:

- the channel separation
- the measurement channel bandwidth
- the adjacent channel bandwidth (if different from measurement channel bandwidth) and
- the center frequency of the reference channel

The measurement is applicable to both modulated and unmodulated signals and provides a means of assessing the transmitters selectivity (Figure 105).

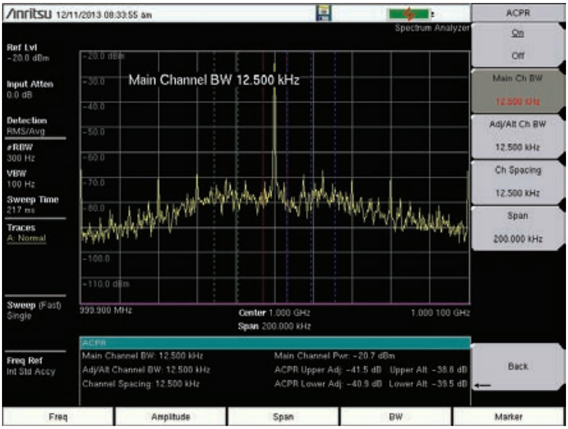


Figure 105

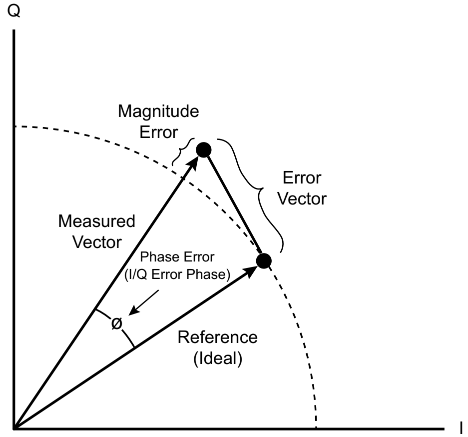
Burst Average Power

Time domain spectrum analysis is a vital tool for analyzing pulsed or burst signals. One important measurement is burst average power which computes the average power within the burst "on" time (Figure 106). Using the same measurement function, the average power within bursts can also be measured.

Figure 106

Error Vector Magnitude (EVM)

The basis of the demodulation measurement on a Vector Signal Analyzer, like MS2830A or MS2692A, is the comparison of a measured signal with a "perfect" ref-erence signal with respect to both magnitude and phase. For each batch of symbols to be analyzed, the demodulated bits of the measured signal are used as the input for creating a perfect reference signal within the analyzer. Note that the EVM can be calculated during the transition between constellation points as well as the sample times when the symbols are determined.



When combined with measurements of other parameters, EVM encapsulates the effects of many different signal distortions. It can therefore help pinpoint such transmission impairments as:

- Phase Noise and Frequency Error
- IQ imbalance, such as gain and phase mismatch, which would result in local oscillator (LO) leakage and unwanted sideband components
- Signal compression (e.g. in the PA) and nonlinearities
- Spurious components

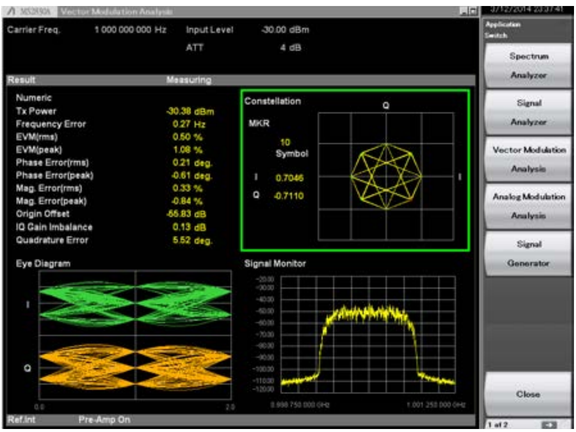


Figure 107: ϖ/4-DQPSK modulation with perfect interference free modulation and ~0% EVM

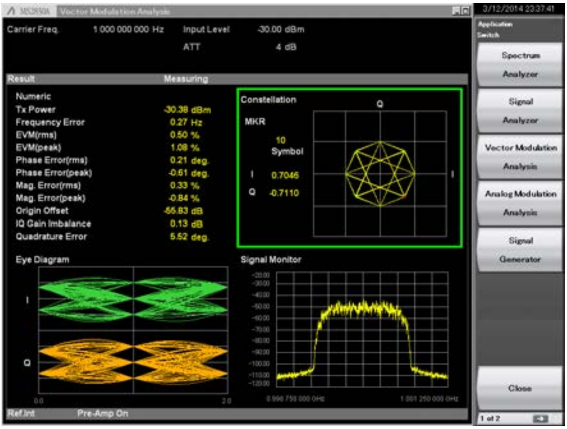


Figure 108: ϖ/4-DQPSK modulation with an AWGN overlay and ~20 % EVM (peak)

It can be seen from this simple diagram above that EVM is influenced by a number of parameters such as below:

- Phase Error
- Frequency Error
- Magnitude Error
- Noise that contributes to all of the above

Each of these areas are contributed to not only by the signal being measured, but also by the test instrument itself which has an effect on how well it can capture the signal, but also how it is able to generate an "ideal" reference signal to use for the calculation. Therefor the limit of EVM demodulation performance can only be as good as the error contributions added during the demodulation process in the signal analyzer.

The average power of the error vector, normalized to signal power, is the EVM. For the percentage format, root mean square (RMS) average is used. The error vector magnitude is equal to the ratio of the power of the error vector to the root mean square (RMS) power of the reference. It is defined in dB as:

$$\text{EVM [dB]} = 10 \log_{10} \left(\frac{P_{\text{Error}}}{P_{\text{Reference}}} \right)$$

where PError is the RMS power of the error vector. For single carrier modulations, Preference is, by convention, the power of the outermost (highest power) point in the reference signal constellation. More recently, for multi-carrier modulations, Preference is defined as the reference constellation average power. EVM is defined as a percentage in a compatible way, e.g. -40 dB equates to 1.0 % EVM.

$$\text{EVM [\%]} = \sqrt{\frac{P_{\text{Error}}}{P_{\text{Reference}}}} \quad 100\%$$

EVM, as conventionally defined for single carrier modulations, is a ratio of a mean power to a peak power. Because the relationship between the peak and mean signal power is dependent on constellation geometry, different constellation types (e.g. 16-QAM and 64-QAM), subject to the same mean level of interference, will report different EVM values. EVM, as defined for multi carrier modulations, is arguably the more satisfactory measurement because it is a ratio of two mean powers and is insensitive to the constellation geometry. In this form, EVM is closely related to Modulation Error Ratio (MER), the ratio of mean signal power to mean error power.

EVM Troubleshooting

Altough EVM is a convenient quantity for specifying performance on a data sheet, analyzing the EVM information is very useful for troubleshooting system perfor-mance. The EVM display and the tabulated vector data on the analyzer provide use-ful information for isolating the cause of system impairments.

Usually EVM troubleshooting is used to depict IQ Modulator Impairments and / or for PA characterization. E.g. for the complete set of IEEE 802.11ac specified transmitter compliance tests the following measurements are mandatory:

- Spectrum Mask
- Spectral Flatness
- Peak Power
- Center Frequency Error
- Symbol Clock Frequency Error
- Center Frequency Leakage
- Error Vector Magnitude (EVM)

EVM versus time

The EVM versus time display can provide useful information, especially in pulsed multiple access or TDMA systems. For example, if the EVM is significantly greater at the start or the end of a pulse, this can indicate timing problems with amplifiers switching on or off too early or too late. High EVM values between symbol times can indicate improper baseband filtering being applied to the signal.

EVM Spectrum

If we apply a Fast Fourier Transform (FFT) to the EVM versus time display, then we will see the distribution of error "energy" with respect to frequency. This can be very useful for revealing the presence of hidden spurs within our signal. A very sharp concentration or error energy in the EVM spectrum reveals the location of the spurious signal, even though it is "buried" within the desired signal itself.

Dynamic EVM

Battery life and power consumption are important considerations for a system-level RF transmitter design. Because the transmit power amplifier (PA) consumes a signif-icant portion of the total system DC power, a number of techniques are employed to reduce PA power usage. Many PAs offer an adjustable DC supply voltage to optimize the maximum RF output power level versus its DC power consumption. Also, most PAs can be powered-down or disabled when not in use to conserve power, such as while receiving or between packets during transmission. In order to maximize power efficiency, the PA must have fast turn-on and turn-off switching times. The highest DC power efficiency occurs when the time delta between PA Enable and the RF signal is minimized, but a short delay can exacerbate transient effects on the RF signal.

Because the power-up/power-down operation of the PA can cause transient and thermal effects that degrade transmitter performance, another metric called Dynamic EVM is often tested. Dynamic EVM is measured with a square wave pulse applied to PA Enable to emulate the actual dynamic operation conditions of the transmitter. The degradation in dynamic EVM is due to the PA transient response affecting the preamble at the start of the packet and causing an imperfect channel estimate. Studies have shown that dynamic EVM with a 50% duty cycle square wave applied to PA Enable to be worse than the static EVM (PA Enable with 100% duty cycle)

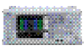
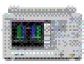
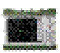
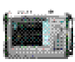
[1] M. Helfenstein, E. Baykal, K. Muller, and A. Lampe. "Error Vector Magnitude (EVM) Measurements for GSM/EDGE Applications Revised under Production Conditions," IEEE International Symposium on Circuits and Systems, May 2005.

[2] M. T. Hunter, A. G. Kourtellis, C. D. Ziomek, and W. B. Mikhael, "Fundamentals of Modern Spectral Analysis," IEEE Instrumentation & Measurement Magazine, August 2011

[3] Allan W. Scott, Rex Frobenius. "RF Measurements for Cellular Phones and Wireless Data Systems", IEEEJohn Wiley & Sons 2008

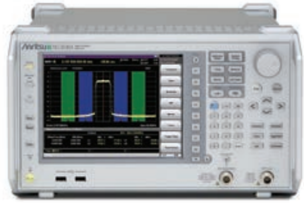
Anritsu Spectrum/Signal Analyzers

Superior performance. Advanced capabilities. Affordable pricing. The Anritsu family of spectrum analyzers/signal analyzers delivers high frequency / level accuracies and a broad set of smart, intuitive features - including models with built-in one button test.

	Model	Frequency Range	RBW	Noise level	Key Features
	**MS2830A RF Models** **MS2830A-040** **MS2830A-041** **MS2830A-043**	9 kHz to 3.6 GHz 9 kHz to 6.0 GHz 9 kHz to 13.5 GHz	Spectrum Analyzer 1 Hz to 3 MHz, 50 kHz, 5, 10, 20*, 31.25 MHz* Signal Analyzer 1 Hz to 1 MHz, 3*, 10 MHz*	Down to -153 dBm** without preamp	• Top-class measurement speed • Supports ±0.3 dB (typ.) total level accuracy • SPA, VSA and Vector SG in one box • Low price, high performance and speed • Excellent SSB phase noise (Opt 066) • Max. 125 MHz analysis bandwidth (Opt) • Supports LTE, LTE-Advanced, GSM/EDGE/EDGE Evo, Mobile WiMAX, WLAN, and W-CDMA etc • Opt wideband preamp
	MS2830A Microwave **MS2830A-044** **MS2830A-045**	9 kHz to 26.5 GHz 9 kHz to 43 GHz	Spectrum Analyzer 1 Hz to 3 MHz, 50kHz, 5, 10, 20*, 31.25* MHz Signal Analyzer 1 Hz to 1 MHz, 3*, 10 MHz*		• 9 kHz to 26.5/43 GHz frequency range; 43 GHz max. built-in pre-amp option • Best-of-class wide dynamic range over 6 GHz • 110 GHz max. frequency range; built-in 1st local signal output for external mixer • For wideband down-converter; built-in 1 GHz IF outputband • Max 125 MHz Analysis bandwidth (Opt) • Opt wideband preamp
	**MS2690A** **MS2691A** **MS2692A**	50 Hz to 6.0 GHz 50 Hz to 13.5 GHz 50 Hz to 26.5 GHz	Spectrum Analyzer 30 Hz to 3 MHz, 50 kHz, 5, 10, 20, 31.25 MHz Signal Analyzer 1 Hz to 1 MHz, 3*, 10 MHz*	Down to -155 dBm** without preamp	• SPA, VSA and Vector SG in one box • Top class performance - Dynamic range 177 dB - Total level accuracy ±0.3 dB (typ.) • Max. 125 MHz analysis bandwidth • High speed modulation analysis • Supports LTE, LTE-Advanced, GSM/EDGE/EDGE Evo, W-CDMA and WLAN etc. • Opt wideband preamp
	**MS2711E** **MS2712E** **MS2713E**	9 kHz to 3 GHz 9 kHz to 4 GHz 9 kHz to 6 GHz	1 Hz to 3 MHz 1 Hz to 3 MHz 1 Hz to 3 MHz	-162 dBm (normalized to 1 Hz) -162 dBm in 1 Hz RBW -162 dBm in 1 Hz RBW	• Handheld, battery-operated design • Lightweight at only 3.5 kg • Dynamic range of > 102 dB in 1 Hz RBW • Phase noise of -100 dBc/Hz max @ 10 kHz offset at 1 GHz
	**MS2720T**	9 kHz to 9 GHz 9 kHz to 13 GHz 9 kHz to 20 GHz 9 kHz to 32 GHz 9 kHz to 43 GHz	1 Hz to 10 MHz	-164 dBm @ 1 Hz RBW -159 dBm @ 1 Hz RBW -160 dBm @ 1 Hz RBW -157 dBm @ 1 Hz RBW -151 dBm @ 1 Hz RBW	• Tracking Generator to 20 GHz • Touch screen operation • Performance to rival benchtop units • Burst detect sweep mode captures emitters as short as 200 μs • Options for all of the major wireless standards • Atomic Clock • Preamp in standard

* with Option **Option and frequency dependent

MS2690A/91A/92A Signal Analyzer



- Frequency coverage up to 6.0 GHz/13.5 GHz/26.5 GHz.
- Total level accuracy: ±0.3 dB (typ.)
- Dynamic range: 177 dB
  TOI: ≥+22 dBm, DANL: -155 dBm/Hz without preamp

Signal Analyzer

- Analysis bandwidth: 31.25 MHz (Std.)/62.5 MHz, 125 MHz(Opt.)
- Modulation analysis software (LTE, LTE-Advanced WiMAX, GSM/GPRS/EDGE, W-CDMA/HSPA/HSPA Evolution, WLAN 11ac/a/b/g/n/p etc.)
- Capture function and Replay function

Vector Signal Generator

- Level accuracy: ±0.5 dB (typ.)
- BER function, Internal AWGN Generator

The Signal Analyzer has the excellent general level accuracy, dynamic range and performance of a high-end spectrum analyzer. Not only can it capture wideband signals but FFT technology supports multifunction signal analyzes in both the time and frequency domains. Moreover, the built-in signal generator function outputs both continuous wave (CW) and modulated signals for use as a reference signal source.

MS2720T Series Handheld Spectrum Analyzer - Spectrum Master™

- Five options offering 9 kHz to 9, 13, 20, 32 & 43 GHz
- Dynamic Range: > 106 dB in 1 Hz RBW @ 2.4 GHz
- DANL: –164 dBm in 1 Hz RBW @ 1 GHz preamp On
- Phase Noise: –112 dBc/Hz @ 10 kHz offset at 1 GHz
- 1 Hz to 10 MHz Resolution Bandwidth (RBW)
- Tracking Generator up to 20 GHz
- Internal Atomic Clock option for the ultimate in handheld frequency accuracy

The highest performance handheld spectrum analyzers available. Providing field technicians and engineers with performance that rivals a benchtop spectrum analyzer. The MS2720T features a touchscreen, full-band tracking generators to 20 GHz, and best-in-class performance for dynamic range, DANL, phase noise, and sweep speed, providing unprecedented levels of spectrum monitoring, hidden signal detection, RF/microwave measurements, and testing of microwave backhauls and cellular signals

MS2830A Signal Analyzer (MS2830A - 040/041/043)

- Frequency coverage up to 3.6 GHz/6 GHz/13.5 GHz.
- Total level accuracy: ±0.3 dB (typ.)
- Dynamic range: 168 dB
  TOI: ≥+15 dBm, DANL: -153 dBm/Hz without preamp
- SSB phase noise*:
  -109 dBc/Hz@1 kHz offset
  -118 dBc/Hz@10 kHz offset
- **Signal Analyzer**
- Analysis bandwidth (Opt): 10/31.25/62.5/125 MHz.
- Modulation analysis software (LTE, LTE-Advanced, WiMAX, GSM/GPRS/EDGE, W-CDMA/HSPA/HSPA Evolution, WLAN 11ac/a/b/g/n/p etc.)
- Capture and Replay function
- **Vector Signal Generator**
- Level accuracy: ±0.5 dB (typ.)
- Internal AWGN Generator (Opt.028)

The MS2830A is a high-speed, high performance, cost-effective Spectrum Analyzer/Signal Analyzer. Not only can it capture wideband signals but FFT technology supports multifunction signal analysis in both the time and frequency domains. Moreover, the built-in signal generator function outputs both continuous wave (CW) and modulated signals for use as a reference signal source. *: with Opt. 066

MS2830A Signal Analyzer Microwave (MS2830A - 044/045)

- Frequency coverage up to 26.5 GHz/43 GHz (up to 110 GHz using external mixer)
- Total level accuracy: ±0.3 dB (typ.)
- Dynamic range (25 GHz): 159 dB
  TOI: +13 dBm, DANL: –146 dBm/Hz without preamp
- SSB phase noise: –115 dBc/Hz@100 kHz offset
- Preamp option up to 43 GHz
- Used as wideband down converter: Built-in IF output function, Frequency: 1875 MHz, Bandwidth: 1 GHz (nom.)
- Signal Analyzer
- Analysis bandwidth (Opt.): 10/31.25/62.5/125 MHz

The MS2830A-044/045 Signal Analyzer includes a spectrum analyzer function for measuring up to 110 GHz using an external mixer based on the 26.5 GHz/43 GHz upper frequency limit. It can be customized to support a range of application-specific measurements.

APPENDIX A

Spectrum Analyzer Conversion Factors

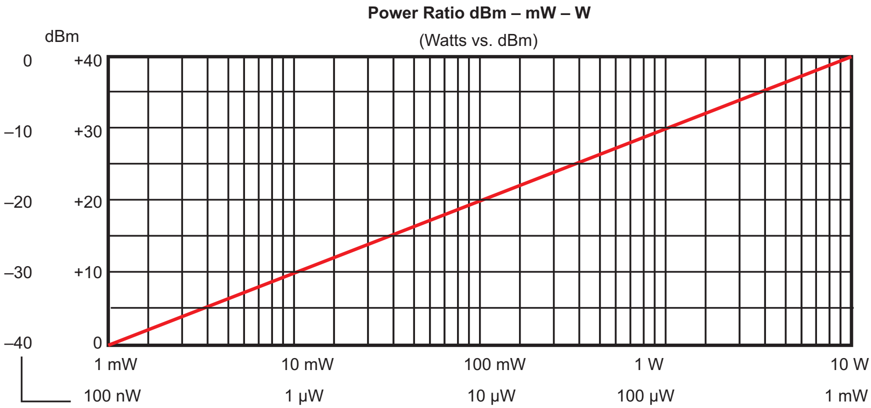
50 Ω Input Inpedance				
To → From →	dBm	dBV	dBmV	dBμV
dBm	0	–13	+47	+107
dBV	+13	0	+60	+120
dBmV	–47	–60	0	+60
dBμV	–107	–120	–60	0

75 Ω Input Inpedance				
To → From →	dBm	dBV	dBmV	dBμV
dBm	0	–11.25	+48.7	+108.7
dBV	+11.25	0	+60	+120
dBmV	–48.75	–60	0	+60
dBμV	–108.75	–120	–60	0

SWR – Reflection Coefficient – Return Loss

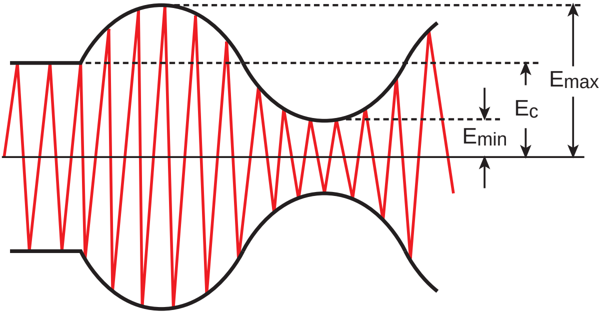
SWR	Reflection Coefficient	Return Loss (dB)	SWR	Reflection Coefficient	Return Loss (dB)
17.391	0.8913	1	1.0580	0.0282	31
8.7242	0.7943	2	1.0515	0.0251	32
5.8480	0.7079	3	1.0485	0.0224	33
4.4194	0.6310	4	1.0407	0.0200	34
3.5698	0.5623	5	1.0362	0.0178	35
3.0095	0.5012	6	1.0322	0.0158	36
2.6146	0.4467	7	1.0287	0.0141	37
2.3229	0.3981	8	1.0255	0.0126	38
2.0999	0.3548	9	1.0227	0.0112	39
1.9250	0.3162	10	1.0202	0.0100	40
1.7849	0.2818	11	1.0180	0.0089	41
1.6709	0.2512	12	1.0160	0.0079	42
1.5769	0.2239	13	1.0143	0.0071	43
1.4985	0.1995	14	1.0127	0.0063	44
1.4326	0.1778	15	1.0113	0.0056	45
1.3767	0.1585	16	1.0101	0.0050	46
1.3290	0.1413	17	1.0090	0.0045	47
1.2880	0.1259	18	1.0080	0.0040	48
1.2528	0.1122	19	1.0071	0.0030	49
1.2222	0.1000	20	1.0063	0.0032	50
1.1957	0.0891	21	1.0057	0.0028	51
1.1726	0.0794	22	1.0050	0.0025	52
1.1524	0.0708	23	1.0045	0.0022	53
1.1347	0.0631	24	1.0040	0.0020	54
1.1192	0.0562	25	1.0036	0.0018	55
1.1055	0.0501	26	1.0032	0.0016	56
1.0935	0.0447	27	1.0028	0.0014	57
1.0829	0.0398	28	1.0025	0.0013	58
1.0736	0.0355	29	1.0022	0.0011	59
1.0653	0.0316	30	1.0020	0.0010	60

Power Measurement

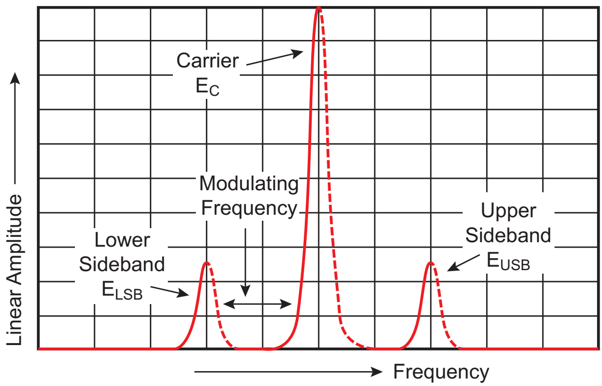


Appendix B

Amplitude Modulation



$$\%M = \frac{(E_{max} - E_{min})}{(E_{max} + E_{min})} \times 100$$

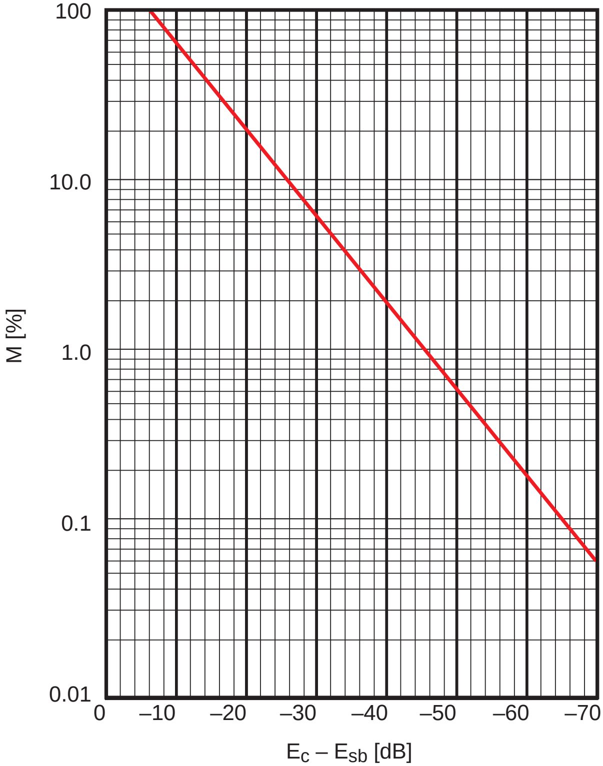


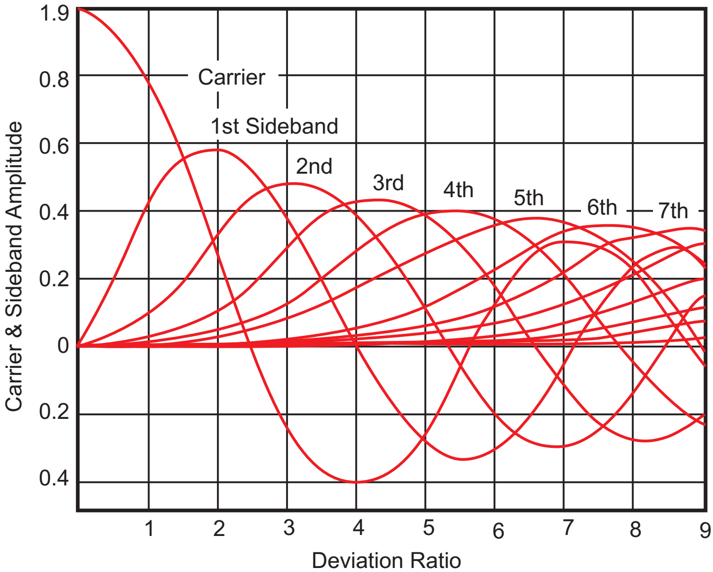
$$\%M = \frac{2E_{LSB}}{E_C} \times 100$$

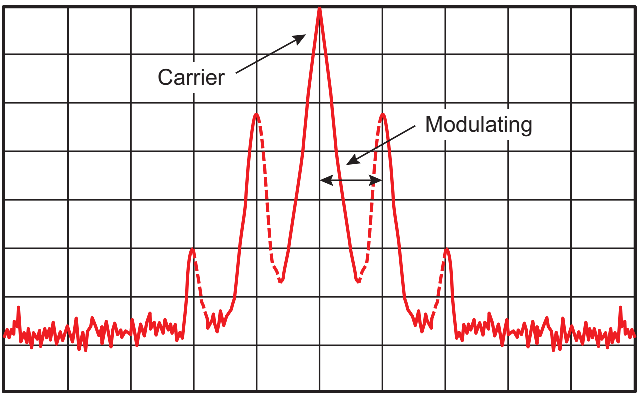
E_c

% Modulation	Side Level Below Carrier (dB)
1	46
2	40
19	26
20	20
30	16.5
40	14
50	12
60	10.4
70	9.1
80	7.9
90	6.9
100	6.0

Sideband Leverl Below Carrier (dB)	% Modulation
10	63
20	20
30	6.3
40	2.0
50	0.63
60	0.2
70	0.063
80	0.02


100
10.0
1.0
0.1
0.01
M [%]
0
–10
–20
–30
–40
–50
–60
–70
$E_c - E_{sb}$ [dB]


1.9
0.8
0.6
0.4
0.2
0
0.2
0.4
Carrier & Sideband Amplitude
Carrier
1st Sideband
2nd
3rd
4th
5th
6th
7th
1
2
3
4
5
6
7
8
9
Deviation Ratio


Carrier
Modulating

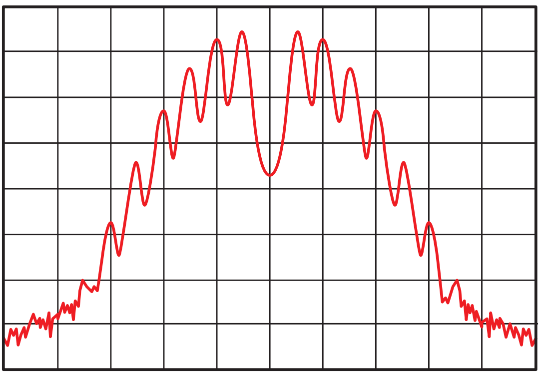
Bessel Functions

Carrier Bassel NULL Number	$M = \Delta F/f$
1st	2.4048
2nd	5.5201
3rd	8.6531
4th	11.7915
5th	14.9309
6th	18.0711
7th	21.2116
8th	24.3525
9th	27.4935
10th	30.6346

Where M = modulation index

ΔF = deviation

f = modulating frequency



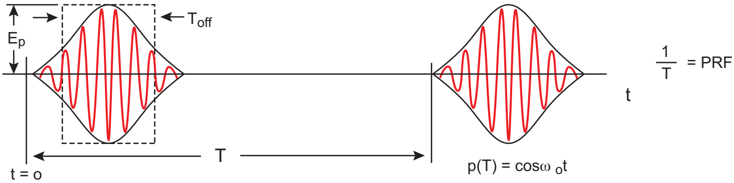
APPENDIX D

Pulse Modulation

$$f_{rep} = \frac{1}{t_{rep}}$$

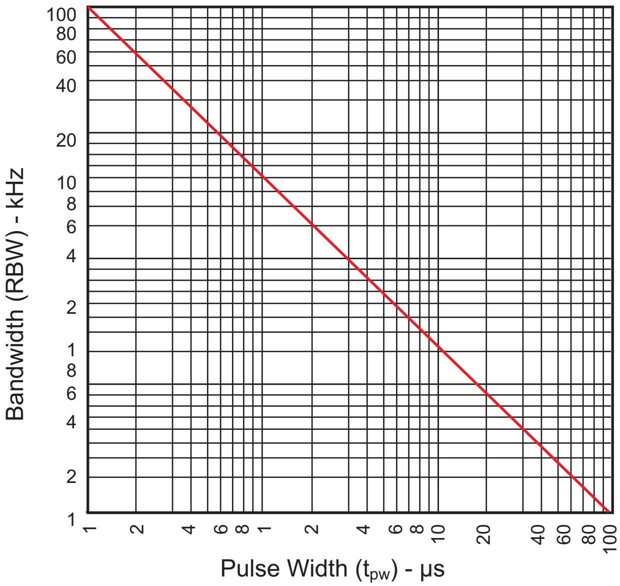
$$t_{rep} = \frac{\text{Sweep time/Div}}{\text{No. of Peaks/Div}}$$

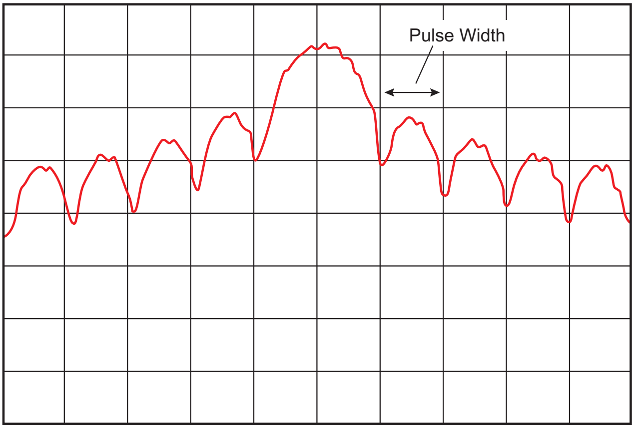
$$t_{pw} = \frac{1}{\text{Sidelobe Width}}$$



T_{eff} = Width of Rectangular Pulse of same height and area as pulse applied to analyzer = $\int_o \frac{p(t)\,dt}{E_p}$

Optimum RBW as a Function of Pulse Width




Pulse Width

APPENDIX E

Intermodulation Distortion / Intercept Points

Calculating Intercept Points requires knowledge of:

1) the order (normally 2nd or 3rd) of the distortion product.

2) input drive level in dBm (example: –30 dBm).

3) the desired or specified suppression of inter-modulation products below the drive level, expressed in dB.

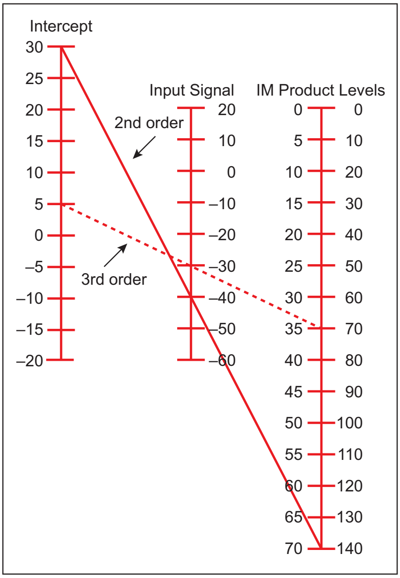
The equation for calculating the intercept point is:

$$I = \frac{\Delta}{(N-1)} + S$$

where: I = intercept point level in dBm for any intermodulation product order.

Δ= suppression of intermodulation products below drive level in dB.

N = order of the intermodulation product.



S = drive level of the input tones (signals) in dBm.

Specifications are subject to change without notice.

● United States
Anritsu Company
1155 East Collins Blvd., Suite 100, Richardson,
TX 75081, U.S.A.
Toll Free: 1-800-267-4878
Phone: +1-972-644-1777
Fax: +1-972-671-1877

● Canada
Anritsu Electronics Ltd.
700 Silver Seven Road, Suite 120, Kanata,
Ontario K2V 1C3, Canada
Phone: +1-613-591-2003
Fax: +1-613-591-1006

● Brazil
Anritsu Eletrônica Ltda.
Praça Amadeu Amaral, 27 - 1 Andar
01327-010 - Bela Vista - São Paulo - SP - Brazil
Phone: +55-11-3283-2511
Fax: +55-11-3288-6940

● Mexico
Anritsu Company, S.A. de C.V.
Av. Ejército Nacional No. 579 Piso 9, Col. Granada
11520 México, D.F., México
Phone: +52-55-1101-2370
Fax: +52-55-5254-3147

● United Kingdom
Anritsu EMEA Ltd.
200 Capability Green, Luton, Bedfordshire, LU1 3LU, U.K.
Phone: +44-1582-433200
Fax: +44-1582-731303

● France
Anritsu S.A.
12 avenue du Québec, Bâtiment Iris 1- Silic 612,
91140 VILLEBON SUR YVETTE, France
Phone: +33-1-60-92-15-50
Fax: +33-1-64-46-10-65

● Germany
Anritsu GmbH
Nemetschek Haus, Konrad-Zuse-Platz 1
81829 München, Germany
Phone: +49-89-442308-0
Fax: +49-89-442308-55

● Italy
Anritsu S.r.l.
Via Elio Vittorini 129, 00144 Roma, Italy
Phone: +39-6-509-9711
Fax: +39-6-502-2425

● Sweden
Anritsu AB
Kistagången 20B, 164 40 KISTA, Sweden
Phone: +46-8-534-707-00
Fax: +46-8-534-707-30

● Finland
Anritsu AB
Teknobulevardi 3-5, FI-01530 VANTAA, Finland
Phone: +358-20-741-8100
Fax: +358-20-741-8111

● Denmark
Anritsu A/S
Kay Fiskers Plads 9, 2300 Copenhagen S, Denmark
Phone: +45-7211-2200
Fax: +45-7211-2210

● Russia
Anritsu EMEA Ltd.
Representation Office in Russia
Tverskaya str. 16/2, bld. 1, 7th floor.
Russia, 125009, Moscow
Phone: +7-495-363-1694
Fax: +7-495-935-8962

● United Arab Emirates
Anritsu EMEA Ltd.
Dubai Liaison Office
P O Box 500413 - Dubai Internet City
Al Thuraya Building, Tower 1, Suit 701, 7th Floor
Dubai, United Arab Emirates
Phone: +971-4-3670352
Fax: +971-4-3688460

● India
Anritsu India Private Limited
2nd & 3rd Floor, #837/1, Binnamangla 1st Stage,
Indiranagar, 100ft Road, Bangalore - 560038, India
Phone: +91-80-4058-1300
Fax: +91-80-4058-1301

● Singapore
Anritsu Pte. Ltd.
11 Chang Charn Road, #04-01, Shriro House
Singapore 159640
Phone: +65-6282-2400
Fax: +65-6282-2533

● P.R. China (Shanghai)
Anritsu (China) Co., Ltd.
Room 2701-2705, Tower A,
New Caohejing International Business Center
No. 391 Gui Ping Road Shanghai, 200233, P.R. China
Phone: +86-21-6237-0898
Fax: +86-21-6237-0899

● P.R. China (Hong Kong)
Anritsu Company Ltd.
Unit 1006-7, 10/F., Greenfield Tower, Concordia Plaza,
No. 1 Science Museum Road, Tsim Sha Tsui East,
Kowloon, Hong Kong, P.R. China
Phone: +852-2301-4980
Fax: +852-2301-3545

● Japan
Anritsu Corporation
8-5, Tamura-cho, Atsugi-shi, Kanagawa, 243-0016 Japan
Phone: +81-46-296-1221
Fax: +81-46-296-1238

● Korea
Anritsu Corporation, Ltd.
5FL, 235 Pangyoyeok-ro, Bundang-gu, Seongnam-si,
Gyeonggi-do, 463-400 Korea,
Phone: +82-31-696-7750
Fax: +82-31-696-7751

● Australia
Anritsu Pty. Ltd.
Unit 21/270 Ferntree Gully Road, Notting Hill,
Victoria 3168, Australia
Phone: +61-3-9558-8177
Fax: +61-3-9558-8255

● Taiwan
Anritsu Company Inc.
7F, No. 316, Sec. 1, NeiHu Rd., Taipei 114, Taiwan
Phone: +886-2-8751-1816
Fax: +886-2-8751-1817

Please Contact:

Issue 2, 09/2013