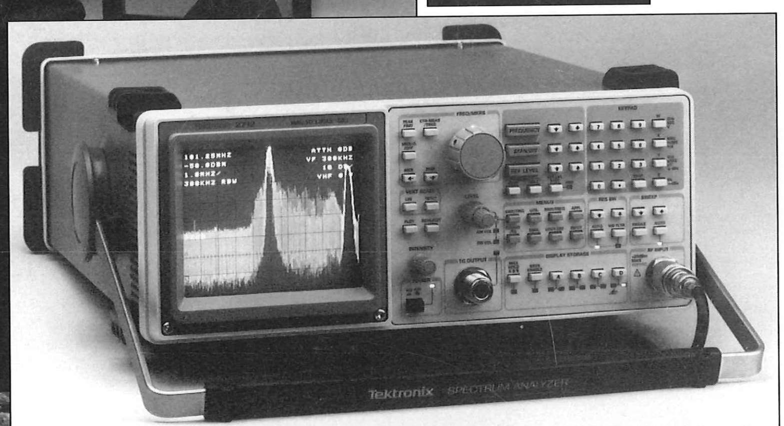
Tektronix

Application Note

Spectrum Analyzer Fundamentals



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Introduction

This application note tells anyone with a basic knowledge of electronics how to make practical measurements with a spectrum analyzer. From it you will learn about the fundamentals of spectrum analysis including:

- Concepts behind spectrum analysis
- Interpreting the spectral display
- Spectrum analyzers and their controls
- Elementary measurement techniques

For best results, use a spectrum analyzer while reading the text, especially when reading the sections dealing with controls and applications. All the photographs in this note were made using Tektronix 49X or 271X series spectrum analyzers. Trying to duplicate the displays, even on a different analyzer, is an effective way to learn the concepts presented here – and gaining a basic understanding of how to use any spectrum analyzer makes it easier to switch from one model to another.

You will need a multi-function RF signal generator to provide the signals used for most of the photos. To carry out the measurements in the tracking generator/spectrum analyzer section you will also need a tracking generator made for use with your analyzer.

This application note doesn't discuss all the controls on all analyzers since some are for special functions, or differ from model to model. You should consult the operator's manual for your analyzer regarding the exact operation of all controls. Other application notes are also available from Tektronix that describe in more detail the operation of spectrum analyzers and techniques for making more advanced measurements.

How to represent electrical signals

When we speak of an *electrical signal*, we generally mean a description of a voltage or current as a function of time. Since we tend to deal more with voltages than currents, we'll use "signal" to indicate a voltage, although current could be used instead.

One way to describe a signal is to plot the voltage as a function of time. The signal can vary in a very structured fashion, it can vary randomly, or it can have both structured and random components.

In the simplest case, that of a *direct current* (DC) signal, specifying the magnitude, or amount, of voltage and its polarity is sufficient because the signal does not change with time. However, when a signal changes with the passage of time, we have to describe not only the amount that it changes, but also how it changes. Signals which change with time are called *time-varying signals.*

Any signal can be represented as a DC voltage plus an *alternating current* (AC) voltage. Because AC signals switch polarity with the passage of time, they can only be time-varying. The DC component, on the other hand, is by definition, constant: it is the amount by which the average value of the AC component is shifted from zero volts. This is shown graphically in Figure 1. Hereafter, we'll deal only with AC signals remembering that we can shift them as needed by simply adding a DC voltage.

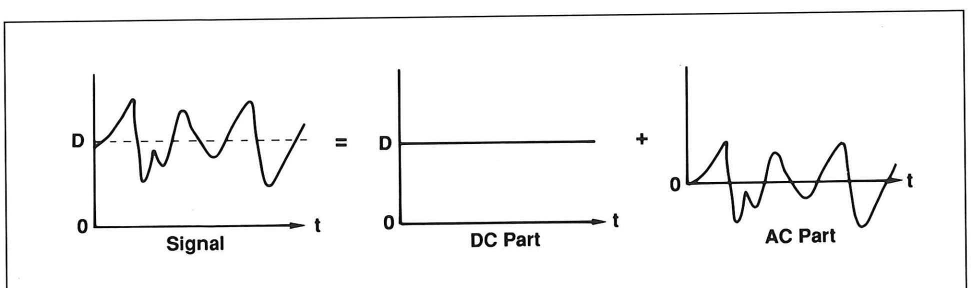


Figure 1. Representing a signal as a DC plus AC part.

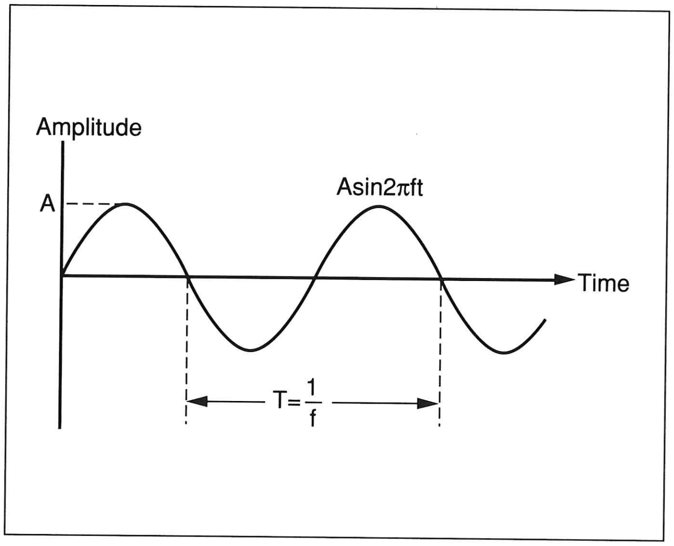


Figure 2A. Time domain drawing of a sine wave.

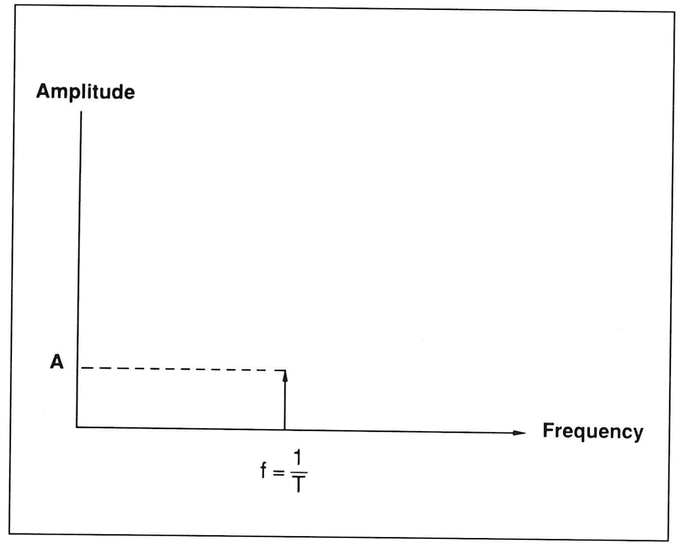


Figure 2B. Theoretical frequency domain representation of the sine wave in Figure 2A.

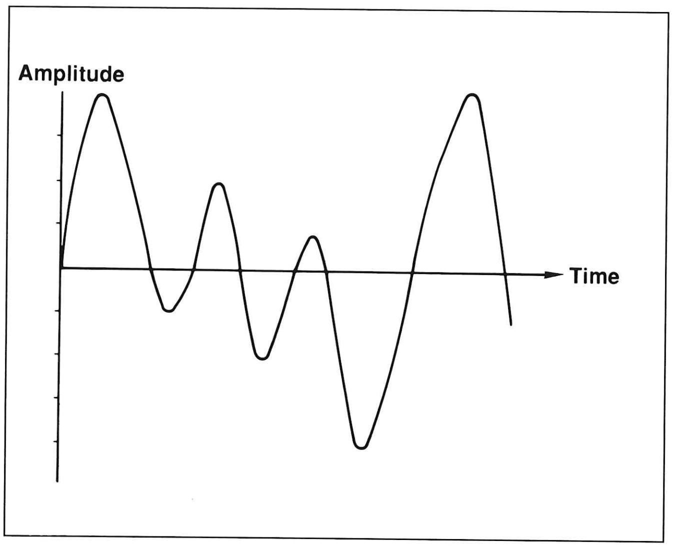


Figure 3A. Time domain sketch of a "complex" signal.

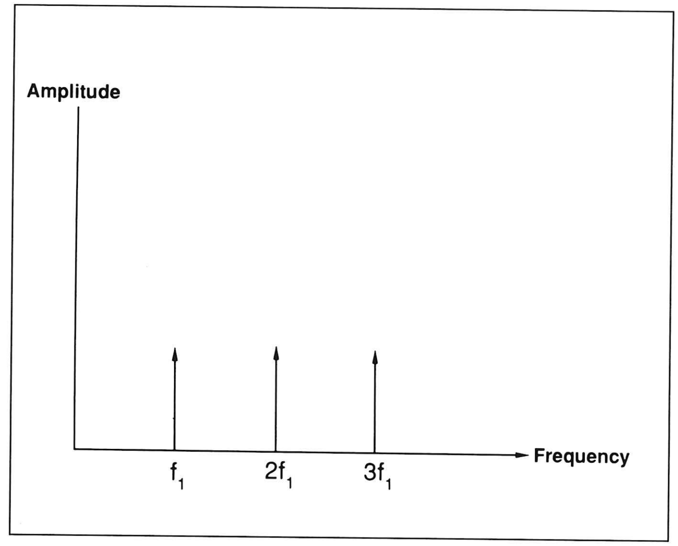


Figure 3B. Frequency domain representation of complex signal from Figure 3A.

The Frequency and Time Domains. Figure 2A shows a signal that varies sinusoidally with time: it is usually called a *sine wave*. In the figure, A is the *amplitude* of the signal. The *frequency* of the sine wave, f, is how many times per second the polarity *oscillates* or switches between plus and minus. The *period* of the sine wave, T, is the reciprocal of the frequency (T = 1/f, one divided by the frequency).

Now here's a curious thing: all sine waves have the same shape. They can differ in amplitude and oscillate faster or slower, but they all look the same. Once you've seen one sine wave, you've seen them all! Therefore, as long as you know what a sine wave looks like, you don't need a mathematical expression, you only need to know the signal is a sine wave of amplitude A and frequency f. So we could represent our sinusoidal signal in another fashion. Figure 2B is a graph of sine wave amplitude vs. sine wave frequency. We see immediately that the signal consists of one sine wave at frequency f with amplitude A. What could be simpler?

The plots in Figures 2A and 2B both represent the same signal. Figure 2A shows it as a function of time; this is called *a time domain representation.* Figure 2B shows the signal as a function of sine wave frequency and is called a *frequency domain representation.* The frequency domain representation is often called the *spectrum* of a signal.

You may think, "This is all well and good, but if I'm looking at sine waves, I don't need any kind of graphical display; all I need to know is the amplitude and frequency. What happens if the signal isn't a sine wave?"

Well, many years ago – and you'll have to take this on faith or learn *Fourier analysis* – it was discovered that a signal can be represented as a sum of simple sine waves. In other words, you could construct a specific signal by adding

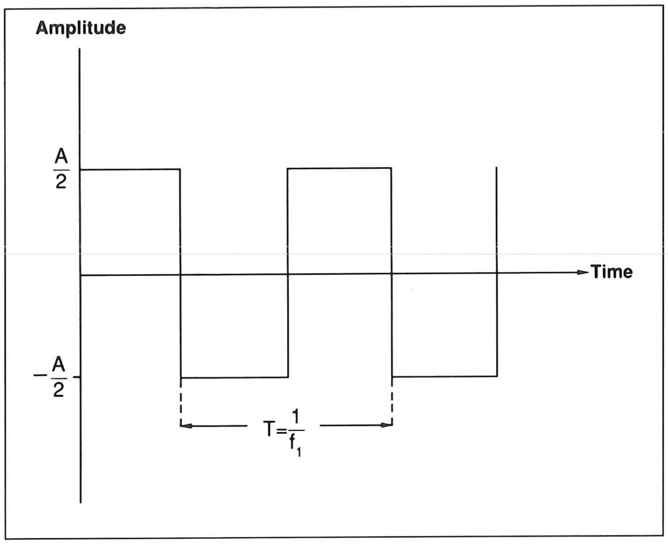


Figure 4A. Time domain drawing of a square wave.

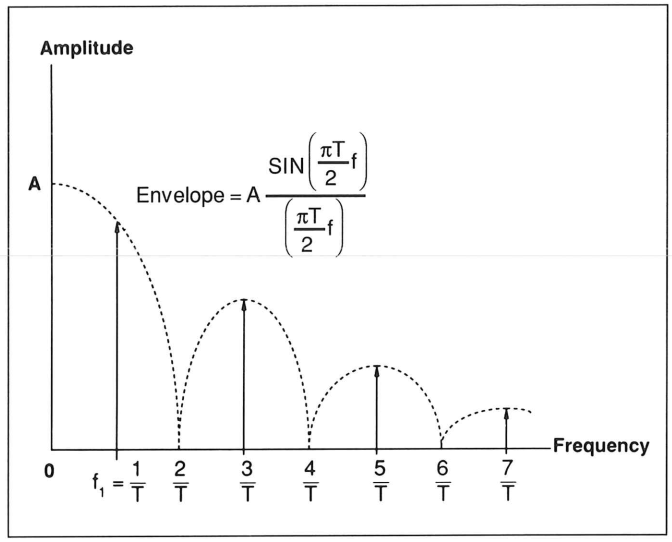


Figure 4B. Theoretical frequency domain representation of the square wave in Figure 4A.

together the outputs of the necessary number of sine wave generators. This can actually be done; it is not just a theoretical proposition! Of course, you would have to correctly adjust the amplitude, phase, and frequency of each generator first.

The reverse of this synthesis process is to take a signal apart to determine which sine wave frequencies are present and what their amplitudes are. This is called *spectrum analysis*. The frequency domain representation, or spectrum, of a signal is a graphical display showing the amplitude and frequency of its sine wave components.

Figure 3A shows a signal that varies with time in a more complex manner than a simple sine wave. It's not apparent that there is any compact way to write a time domain representation for this signal; you might even think it's random. However, when we look at its spectrum, which is shown in Figure 3B, we see that the signal is, in fact, just the sum of three sine waves of the same amplitude at multiples of the same frequency. Our "complex" signal has become a relatively simple one! (You can verify this example for yourself: draw 3 equal amplitude in-phase sine waves at multiples of 1, 2, and 3 times the same frequency on graph paper and add them together point by point.)

As another example, Figure 4A shows the time domain representation of a simple *square wave*. A square wave is a special case of a *rectangular pulse train* in which the signal level remains high exactly as long as it is low. The amplitude of the square wave is A/2 and its pulse repetition rate is 1/T (T is the period of the square wave). The repetition rate is called the *fundamental frequency*.

Figure 4B shows the frequency domain representation of the same square wave. Because the signal components occur at multiples of the fundamental frequency, they are called *harmonics*; the second harmonic is twice the frequency of the fundamental, the third harmonic frequency is three times that of the fundamental, and so on. In this case, the frequency components theoretically go on forever. The amplitudes of the components of any rectangular pulse train are limited by a (sin x)/x function which is shown as a dashed line in Figure 4B. The (sin x)/x function is referred to as the *envelope* of the spectrum and the humps in the envelope are called *lobes*. In the special case of a square wave, all even harmonics are zero because they occur at the zeroes of the (sin x)/x function. The odd harmonics occur at the peaks of the (sin x)/x function. Their amplitudes are equal to $(2A/\pi)$, $(2A/3\pi)$, $(2A/5\pi)$, $(2A/7\pi)$, etc.

Amplitude Factors. Thus far, we've discussed signals as though they are simply voltages. That is, a voltage which is twice another appears to be twice as large on the display screen. This approach is appealing and useful in the time domain when the instantaneous amplitude of a single signal is examined. In the frequency domain, when multiple signals of greatly different amplitudes are measured, it doesn't necessarily make best use of available display space. Consider that many CRT *graticules* (the calibrated grid pattern overlaying the screen) are divided vertically into eight major divisions. Each division is further subdivided into five minor divisions. Thus, a signal of one minor division in amplitude can be measured on the same scale as another signal eight major divisions in amplitude. The ratio of smallest to largest signal that can be reasonably measured is then 40 to 1.

One of the very desirable features of spectrum analyzers is that they are capable of simultaneously measuring voltages which differ by factors of 10,000 or more. To display signal voltages which differ by only 1,000 to 1 would require a graticule with 200 divisions! How can such a wide range of voltages be displayed? The answer is the decibel.

The *decibel*, or dB, is equal to ten times the logarithm of the ratio of one power to another. If one power is twice another power, it is three decibels (+3 dB) greater(if it were half, it would be –3 dB). For every doubling or halving of the power, the decibel value changes by three.

Because power is proportional to the rms voltage squared divided by the resistance, we can also define the dB as twenty times the logarithm of the ratio of one voltage to another if the resistances across which they are measured are the same. If one voltage is twice another voltage, it is six decibels (6 dB) greater (if it were half, it would be –6 dB). For every doubling or halving of the voltage, the decibel value changes by six. Note that because a dB is a dB regardless of whether power or voltage is measured, doubling the voltage is equivalent to quadrupling the power (6 dB = 3 dB + 3 dB).

So how does the dB help us display measurements that are widely different? Suppose that instead of letting each division on the CRT graticule equal a voltage range, we let each division represent decibels. At 10 dB per division we can now display 80 dB if the graticule has 8 divisions. This represents a voltage range of 10,000 to 1.

Suppose we choose to measure power ratios with a reference power of 1 milliwatt. The reference unit is then said to be the dBm, or dB with respect to 1 milliwatt. Equivalently, if we select a reference voltage of 0.224 rms volts measured across 50 ohms, the units remain dBm (because 0.224 squared and divided by 50 equals one milliwatt).

Other "types" of dBs are possible. For instance, if the reference is one volt, the unit is dBV. Similarly, if the reference is one millivolt, the unit is dBmV. The CATV industry routinely uses the dBmV and measures the voltages across a resistance of 75 ohms. Regardless of the reference unit selected, all subsequent measurements must be made in the same units across the same resistance.

Voltages are usually measured on a linear scale, whereas decibels represent a logarithmic scale. On a linear scale, each increment represents a **fixed difference** between signal levels; on a logarithmic scale, each increment represents a **fixed ratio** between signal levels. It is important to understand this difference. On a linear scale with eight divisions, a voltage six divisions high is three quarters as large as one which is eight divisions high; a voltage four divisions high is half of one eight divisions. On a log scale with eight divisions of 10 dB/division, a voltage six divisions high is only one tenth as large as one which is eight divisions high. Because the ratio is constant, any signal that is 20 dB less than another is one tenth as large. Therefore, a voltage four divisions high is also only one tenth as large as one which is six divisions high. In a similar way, any signal that is 40 dB less than another, is one hundredth as large, so that a voltage four divisions high is only one hundredth as large as one which is eight divisions high.

An obvious problem with a scale that allows such a large range of signals on screen simultaneously is that two signals that appear close in amplitude may actually differ significantly. As an example, assume there are two signals that differ by 3 dB. If the scale factor is 10 dB/div, they are separated by only 0.3 divisions. This is less than 4% of the scale even though one signal has twice the power of the other.

To allow better resolution of signals with similar amplitudes, spectrum analyzers let you switch to a smaller scale factor, say 2 dB/div. In the example of two signals 3 dB apart, the display would then show them separated by 1.5 divisions.

Some instruments also allow you to switch to a linear display mode to directly read the value of the signal voltage. The rms signal voltage is displayed with a scale factor of volts/div, millivolts/div, etc.; zero is at the bottom of the screen.

What is a Spectrum Analyzer?

A *spectrum analyzer* is a device for measuring and displaying the frequency domain representation, or spectrum, of a signal. There are different types of spectrum analyzers, but in this note we will be concerned only with *heterodyne spectrum analyzers* (sometimes called scanning spectrum analyzers). A heterodyne spectrum analyzer is essentially a radio receiver. Try this: turn on a conventional broadcast receiver equipped with a signal strength meter and plot a graph of the meter reading versus the indicated frequency as you tune from one end of the band to the other. The graph is the spectrum of the broadcast band. The spectrum tells you at which frequencies the signals occur and how strong they are. In order to plot the graph, you had to decide on a horizontal scale factor of so many kHz or MHz per division; this is called the *span/division.* A scanning spectrum analyzer produces a similar display on its CRT.

In this example, you manually tuned, or *scanned* the broadcast band. Some of the stations may have been so close together that you heard them simultaneously, and could not get an independent meter reading for each. This happens when the *resolution bandwidth (RBW)* of the receiver is too wide to *resolve*, or separate, the stations. In this case, the RBW equals the *intermediate frequency (IF)* bandwidth of the receiver. When you stop tuning, the receiver is no longer scanning; it is in zero span mode. The receiver output is now a time domain representation of the amplitude modulation on the signal coming through the IF filter at the frequency to which the receiver is presently tuned. The detected signal appears at the receiver's speaker as the sound you hear.

A spectrum analyzer operates similarly, but there are important differences. The spectrum analyzer can accommodate a wider range of signal frequencies than your typical broadcast receiver. You have to tell the analyzer how wide a band to scan and where the band is located. You do this by setting the span/division control and adjusting the center or start frequency of the display. The scan is then performed automatically. In order to carry out measurements at extremely high frequencies, some analyzers make use of *band switching* in much the same way that communications receivers do to listen to a wide range of communications frequencies. Your receiver has an automatic gain control to regulate signal levels, but with a spectrum analyzer, you control the signal level manually. Care must be exercised; too large a signal can cause erroneous results by overloading portions of the analyzer's circuitry, while too much attenuation may result in the signal "getting lost in the noise." The spectrum analyzer also has a selection of resolution bandwidths to choose from. Multiple RBWs are needed because in some cases you must separate closely-spaced, narrowband signals, while in others you must examine signals with larger bandwidths.

There is a maximum speed at which a band can be accurately scanned with a RBW of a given width; generally the smaller the RBW, the slower the speed. Modern spectrum analyzers automatically select the optimum speed for you.

What Does a Spectrum Analyzer Do? Spectrum analyzers measure and display the sine-wave frequency components of signals present at their inputs. Consequently, applications for spectrum analyzers include:

- Examining signal levels, bandwidths, and frequencies
- Determining noise power and carrier-to-noise ratios
- Measuring distortion (intermodulation and harmonic), percent modulation, and FM frequency deviation
- Detecting spurious signals
- Aligning transmitters and receivers
- Checking specifications

With the addition of a tracking generator, filter bandwidth, amplifier frequency response, and standing wave ratios (SWR) can also be checked. Many of these measurements are described in this application note or other Tektronix publications.

Spectrum Analyzer Controls

Most modern spectrum analyzers employ three primary controls:

- Reference level
- Frequency
- Span per division

It is possible to make a variety of measurements using only the primary controls, but additional controls are also provided. The added controls not only make using the analyzer more convenient, but also make the analyzer more adaptable to your measurement needs. Many features of modern spectrum analyzers are microprocessor controlled and selectable from on-screen menus. The photograph on pages 20 and 21 shows the control panel of a Tektronix 2712 Spectrum Analyzer.

Reference Level Control. The *reference level control* varies the level of the signal necessary to produce a full-screen deflection. The *reference level* is the signal strength needed to deflect the CRT display to the top line of the graticule. If the reference level is set for –10 dBm, a –10 dBm signal would rise just to the top graticule line. To determine the level of any other signal, multiply the number of divisions it is below the reference line by the vertical scale factor and subtract the result from the reference level. If the reference level is –10 dBm and the scale factor is 10 dB per division, a signal two divisions down from the top is a –30 dBm signal (–10 dBm ref lvl – 2 div x 10 dB/div = –30 dBm).

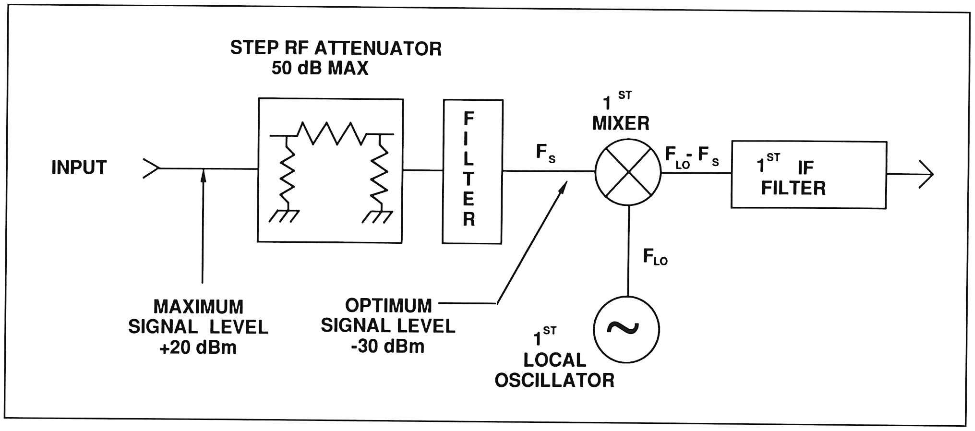


Figure 5. Spectrum analyzer input stages indicating optimum 1st mixer level and frequency mixing.

Tektronix spectrum analyzers display both the reference level and scale factor on screen (see pages 20 and 21).

The reference level is determined by the RF attenuation and the IF gain, but attenuation and gain are controlled by independent sections of the analyzer. To avoid having to operate two controls, most analyzers, like the Tektronix 49X, 271X, and 275X series, automatically select the proper amounts of RF attenuation and IF gain. Therefore, you operate only one control to set the reference level.

The *RF attenuator* determines the amount of attenuation the signal encounters just after it enters the analyzer. **For optimum analyzer performance, the input signal must be attenuated to a level at the 1st mixer specified by the manufacturer**. A *mixer* is a frequency shifting circuit. In modern analyzers, there are usually three such frequency shifting circuits. The level of signals entering the 1st mixer is critical to optimum analyzer performance. Exceeding the specified *1st mixer input level* can result in distortion and spurious signal products, or, in extreme cases, damage to the mixer.

Tektronix 271X series spectrum analyzers have an optimum signal level at the input to the 1st mixer of –30 dBm.

Figure 5 demonstrates how the RF attenuator reduces the input signal amplitude to the optimum level at the 1st mixer input. For instance, if the signal at the analyzer input is –10 dBm, the RF attenuator should be set for 20 dB of attenuation for the 1st mixer to see a signal level of –30 dBm.

The *IF gain* is then selected to control the amount of amplification following the 1st mixer to keep the analyzer within amplitude calibration.

All analyzers have a *maximum input level* that must not be exceeded. Typically, this level is +20 to +30 dBm (2.24 to 7.07 volts across 50 ohms). Spectrum analyzers are also susceptible to damage from DC voltages. This is extremely important to remember. If a DC voltage can be safely applied to an analyzer, the maximum value is usually indicated on the front panel near the input connector. If DC voltage is not permitted, use an external *blocking capacitor* when measuring signals with a DC component.

When applying signals with DC or very low frequency components to the analyzer, here is the preferred method:

- Set the analyzer reference level to maximum
- If your analyzer cannot handle DC voltages, install an external blocking capacitor
- If the voltage is very large, install an external attenuator ahead of the analyzer
- Connect the signal to the analyzer
- Reduce internal attenuation to achieve satisfactory display of desired signal

The maximum input level specification applies to the sum of all signals present at the input (except DC), regardless of whether they are displayed on the screen or not – **what you don't see can hurt you!** For instance, if two signals of +18 dBm are present, the input circuitry is actually being exposed to at least +21 dBm. Two equal amplitude signals result in a combined level 3 dB higher than either alone. This exceeds the maximum input level of some analyzers.

Caution

Exceeding the maximum input level specification of your analyzer can cause extensive and expensive damage to your instrument.

As another example, suppose signals of +10 dBm and –60 dBm are present at the input. You need 40 dB of RF attenuation to reduce the +10 dBm signal to –30 dBm, the optimum level at the input to the 1st mixer. If this makes the reference level +10 dBm, the larger signal peak will be at the reference level and the smaller peak will be seven divisions down at a scale factor of 10 dB/div. You will be unable to get a good look at the smaller signal. Removing RF attenuation to decrease the reference level – even if the larger signal is off screen to the left or right – would increase the level of the larger signal at the input to the 1st mixer to more than –30 dBm. However, IF gain may be added to increase the displayed level of the smaller signal, or, in the case of the Tektronix 271X, the 1st mixer input level can be increased to –20 dBm.

Here are several tips for safely using the spectrum analyzer when you are unsure of the amplitude of the input signal. First, always select the largest RF attenuation or maximum reference level available, and use the widest span to ensure that signals which otherwise might be off screen are clearly visible. Use external attenuation if the total signal level can exceed the input specification of your analyzer. Tentatively bring the signal cable into contact with the analyzers input connector while watching the display. If no unduly large signals appear, finish connecting the cable. After the cable is safely connected and the signal is displayed on the screen, the reference level can be reduced a step at a time to bring the largest signal to the top of the screen. Second, if an RF power meter is available, use it to check the signal level of the source before connecting the source to your analyzer. Be certain that the RF power meter measures the total signal power from all signals present at the input to the meter. Last, check DC and AC signal levels before connecting the analyzer using an oscilloscope, voltmeter, RF voltmeter, or other instrument.

Frequency Control. Scanning spectrum analyzers use a series of local oscillators and mixing circuits to measure the spectrum of the input signal. The first local oscillator (1st LO) determines the range of input frequencies analyzed. Figure 5 shows a typical analyzer front end. By sweeping the frequency of the 1st LO over a specified frequency range or span, a corresponding range of input signal frequencies is swept past the resolution bandwidth (RBW) filter. The *frequency control* customarily determines the center of the swept frequency range and makes that point the center of the screen. The frequency is then referred to as the *center frequency*. However, in some modes of operation the frequency control determines the starting point of the frequency sweep and references it to the left edge of the screen. The reference point is then called the *starting frequency*. Tektronix analyzers read out the center or starting frequency at the top of the screen. When display storage is used on Tektronix analyzers, there is also an intensified spot on the waveform indicating the point specified by the frequency control.

Most spectrum analyzers use a large knob on the front panel for frequency control. For instance, turning the Frequency/Markers knob on a 2711 or 2712 clockwise increases the center or starting frequency, whereas turning it counterclockwise decreases the frequency.

However, digital technology enables most modern analyzers to provide additional modes of frequency control, avoiding the necessity of manually tuning through the whole frequency band, and making many measurements more convenient to carry out. For example, on the 2712 you can increase or decrease the frequency by pressing the up or down frequency arrow keys, or you can press the FREQUENCY key and enter the desired frequency directly from the keypad (this can also be done via the Marker/Frequency Menu on the 2711). In addition, tabular and programmed tuning modes are available for more specialized applications.

Span Control. The *span control* or *span/div control* regulates the width of the frequency spectrum that is displayed by controlling the width of the local oscillator sweep. You set the number of kHz or MHz represented by a division of the horizontal axis by operating the span/div control. Because the frequency axis is usually ten divisions long, the span/div control also determines the total frequency span. Thus, with 20 MHz selected as the span/div, a 10 division screen sweeps across a spectrum of 10 div x 20 MHz/div, or 200 MHz. If the center frequency was set for 175 MHz, then the analyzer would sweep from 75 MHz to 275 MHz. Span/div can be set via the span arrow keys in a 1-2-5 sequence on the 2711 or 2712, or, on the 2712, by pressing the SPAN key and then entering the desired value from the keypad (this can also be done via the Marker/Frequency Menu on the 2711). The selected span/div is read out on screen.

Span/div controls usually have two settings that do not require you to specify the span in Hertz/div. These are the *MAX SPAN* and the *ZERO SPAN* modes. You can toggle the 271X in and out of these modes using the corresponding front-panel key.

In MAX SPAN, the analyzer sweeps across its maximum frequency range. If the maximum input range of your analyzer is 0 Hz to 1800 MHz, then the analyzer sweeps the frequency spectrum from 0 Hz to 1800 MHz when in max span.

In ZERO SPAN, the analyzer does not sweep across a frequency range. Instead it behaves like a conventional (superheterodyne) radio receiver. The analyzer is "tuned" to the center

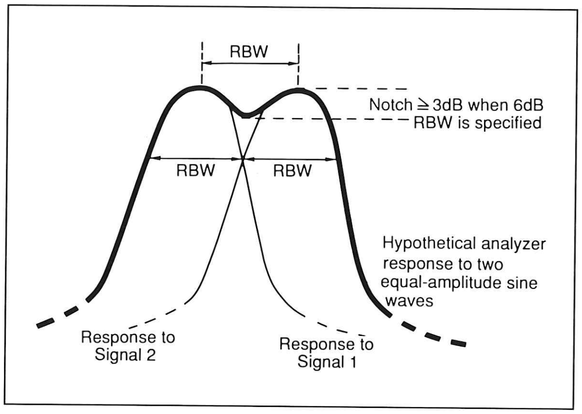


Figure 6. Two closely spaced sine waves "just resolved".

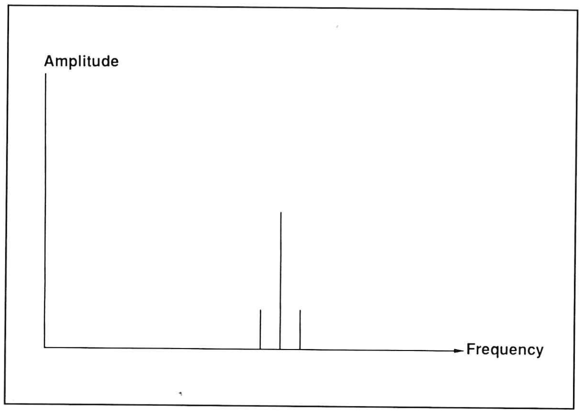


Figure 7. Simple spectrum drawn with fine-tipped pen reveals closely-spaced signals.

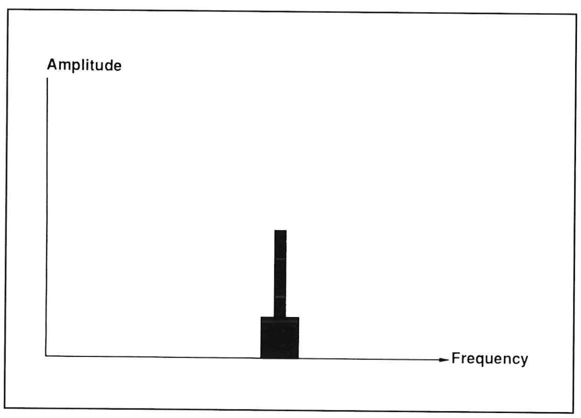


Figure 8. Simple spectrum drawn with broad-tipped pen obscures closely-spaced signals.

frequency and the signal present in the RBW filter passband is continuously displayed. If, for instance, your analyzer has a speaker attached to its output (standard equipment on the 271X) and you center the analyzer on a local AM radio station using a RBW filter at least 3 kHz wide, you would actually hear the program material. The analyzer now basically works like an oscilloscope attached to a receiver. The horizontal scale changes from frequency/division to time/division.

Resolution Bandwidth (RBW) Control. *Resolution bandwidth (RBW)* filters are bandpass filters located (usually) in the spectrum analyzer's third intermediate frequency stage. They determine how well closely-spaced signals can be separated. The narrower the RBW filter, the closer two signals can be and yet still be seen as separate signals. The RBW filters also determine the analyzers response to pulse-like signals and to noise. The *resolution bandwidth control* selects which RBW filter is used.

The shape of a spectrum displayed on the analyzer screen is a combination of the shape of the RBW filter and the shape of the true signal spectrum. Thus, the measured analyzer response to two **equal-amplitude** sine wave signals that are one RBW apart in frequency resembles Figure 6. The responses due to each signal combine to produce a measured response that is larger than either alone.

The ability of a filter to separate, or resolve, closely spaced signals is dependent on its bandwidth and its shape. The Institute of Electrical and Electronic Engineers (IEEE) defines a spectrum analyzers resolution as "... the ability to display adjacent responses discretely. The measure of resolution is the frequency separation of two responses which merge with a 3 dB notch." The International Electrotechnical Committee (IEC) uses a similar definition.

RBW filters are defined by their bandwidths and shape factors. The width is specified in Hertz either 3 dB or 6 dB down from the filter peak. The 6 dB bandwidth meets or exceeds the IEEE definition of resolution regardless of detector type used in the analyzer when the equal-amplitude sine waves are separated by one RBW; the 3 dB specification is the more familiar "half-power" bandwidth. In addition, international CISPR standards for measuring electro-magnetic interference (EMI) are based on the 6 dB response which approximates the impulse bandwidth of the filter.

The steepness of a filter is indicated by its shape factor. The *shape factor* is the ratio of the RBW filter bandwidth 60 dB down from peak to its nominal bandwidth – generally the smaller the ratio, the sharper the filter. Tektronix analyzers have typical shape factors of 7.5:1 or less. Shape factors are important in determining how far two signals which are **not of equal amplitude** must be separated to be resolved.

Ideally, the RBW filters should be extremely narrow in order to faithfully trace out signal spectral shapes and resolve very closely spaced signals. Note the low level pair of sig-

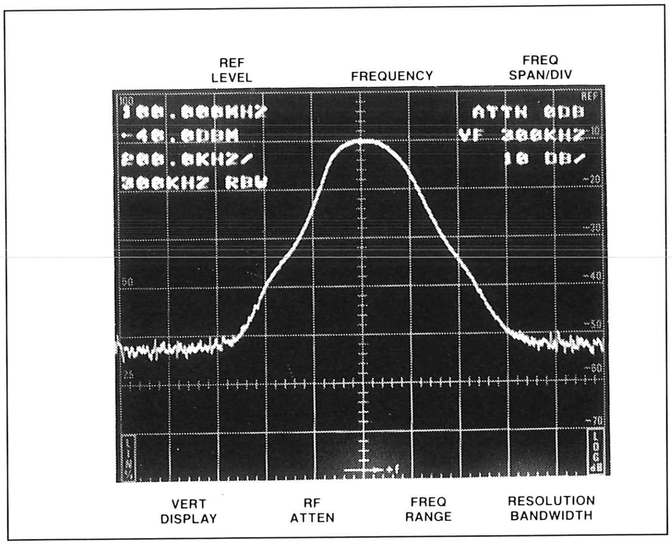


Figure 9. RBW that is too wide obscures nearby signals.

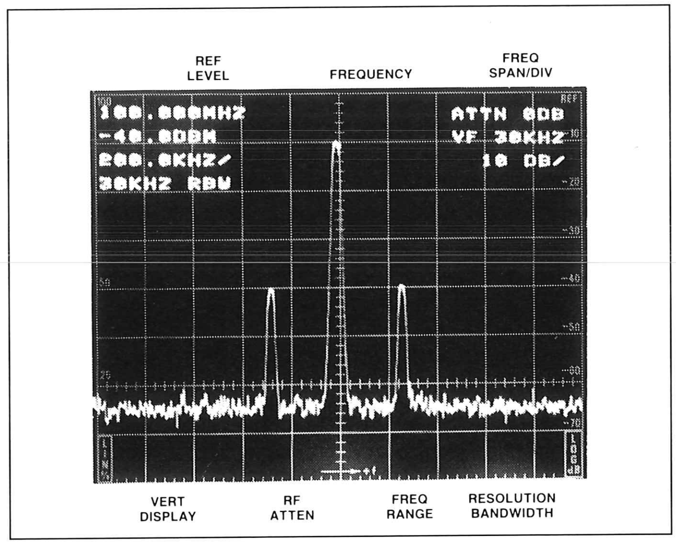


Figure 10. Analyzer-selected RBW reveals closely-spaced signals.

nals bracketing the large center signal in Figure 7. If a wider pen had been used to draw the spectrum, as in Figure 8, the smaller signals could be easily overlooked since they are indicated only by the slight bulge near the bottom of the large center signal.

Increasing the resolution bandwidth produces results much the same as broadening the width of the pen. Figures 9 and 10 demonstrate the results of using RBW filters that are too wide. In each figure, all parameters are the same except the resolution bandwidth. Figure 9 was taken with a RBW filter that was too wide. Figure 10 uses a narrower, analyzer-selected RBW. You can clearly see closely-spaced signals that were not visible in Figure 9.

Why not use as narrow a RBW filter as possible? The primary answer is that the rate the analyzer sweeps through the selected frequency span must be slow enough to allow the signals passing through the filters to reach their peak amplitudes or inaccurate measurements result. The narrower the filter, the longer it takes for the signal to reach its peak value. Therefore, using a narrow RBW filter with a wide span results in a sweep taking an extremely long time. To keep sweep rates reasonably fast, the resolution bandwidth must increase as the span/div increases. Unless a special requirement dictates otherwise, the resolution bandwidth should remain between 1/50 and 1/10 times the span/div.

Another characteristic associated with RBW filters, is that as the bandwidth is narrowed, the displayed *noise floor* decreases. The noise floor is the baseline or lowest horizontal part of the trace. Because of its appearance, this part of the signal is sometimes referred to as "grass". The noise floor (or self-noise) of the spectrum analyzer is the noise that is displayed when no signal is connected to the analyzer. Because you can't see signals lower in amplitude than this noise, the noise floor determines the ultimate sensitivity of the analyzer. The noise floor decreases as the RBW is narrowed because noise power is proportional to bandwidth.

It doesn't matter whether noise is generated internally by the analyzer itself, or whether it accompanies the input signal. When we change the bandwidth of the RBW filter by a factor of ten, the noise floor should change by 10 dB. That is, if the RBW is decreased ten times (e.g. from 300 kHz to 30 kHz), the noise floor should decrease by about 10 dB; if the bandwidth is increased from 300 kHz to 5 MHz, the noise should increase by about 12 dB. If RBW filters could be made in the ideal rectangular shape, the changes would be exact. Because they can't, the amount of noise passed by a filter is not quite the same as predicted from its 3 dB or 6 dB bandwidth. When actually measuring noise power, a correction must be made for this.

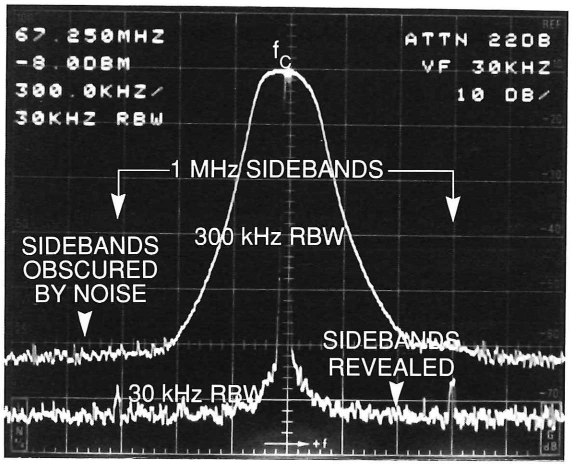


Figure 11. Composite photo showing sidebands obscured by noise when using wider RBW.

The reduction in the noise floor works in our favor when we are looking for low level narrowband signals. Figure 11 is a composite photo of the spectra produced by the same signal when different RBW's are used. In the upper trace, the signal's sidebands are buried in the noise. The RBW was reduced by a factor of ten between the upper and lower traces which, in effect, reduced the noise by 10 dB to reveal the sidebands.

The limitations imposed on a spectrum analyzer by the resolution bandwidth filter are significant. Through the use of microprocessors, modern spectrum analyzers automatically choose the best resolution bandwidth as a function of the span/div and sweep rate selected. AUTO is the normal setting for the RBW control on the 271X. Nevertheless, there will be times such as when analyzing pulse type signals like radar, when manual control of this function is desired.

Sweep Control. The *sweep control* selects the *sweep rate* at which the spectrum is swept and displayed. Sweep rate units are time/division; a typical value might be 20 msec/div. As with resolution bandwidth, most analyzers can automatically select the optimum sweep speed, depending on other parameters (span, RBW, and video filter BW). On the 271X, sweep rate can be varied from the front panel or via the Sweep/Trigger Menu in a 1-2-5 sequence, but AUTO is the normal setting for the sweep control.

If you manually select the sweep rate, remember these pointers.

- When the spectrum is swept too fast, the RBW filter won't have time to charge, and inaccurate measurements will result.
- When swept too slowly, the display accuracy is not effected, but the display may flicker objectionably or fade out entirely before the next sweep is begun. Flicker and fade out can be overcome using display storage.

As described previously, whenever the analyzer is set to ZERO SPAN, it displays a time domain representation of the envelope of the signal passing through the RBW filter at the selected center frequency. The sweep control then functions just like an oscilloscope's sweep rate or time base control; you horizontally expand or compress the displayed time domain signal by changing the sweep rate.

Video Filter Control. A *video filter* is a post-detection filter (sometimes referred to as a *noise averaging* filter) used primarily to reduce noise in the displayed spectrum to its average value and, hence, enhance the visibility of low level signals. Care should be exercised in its use, however, because it can also reduce the indicated amplitudes of certain types of signals such as video modulation and short duration pulses. Most analyzers provide several video filter bandwidths. Currently, convention suggests a default video filter bandwidth equal to the RBW. This is a good starting point for most measurements, and is the value used by late model 2711s and all 2712s. The video filter bandwidth is displayed at the top right of the screen.

The video filter key enables you to toggle an alternate video filter on or off. Normally, the analyzer automatically selects an alternate video filter bandwidth of approximately 1/100 of the RBW, which does a good job of reducing post-detection noise. However, Utility Menu/2/5/1 can be used to specify any of the installed filter widths for the alternate filter. Utility Menu/2/5/0 restores AUTO filter selection. When analyzing pulsed or broadband signals, a narrow video filter should not be used since the low-pass characteristic of the filter doesn't allow the amplitude of the analyzed signals to reach full height (for optimum pulse response, the video bandwidth should be at least three times the 6 dB resolution bandwidth).

Just as with the RBW filter, a signal needs time to reach its peak amplitude when propagating through a video filter. Therefore, the sweep rate should be decreased when the video filter is turned on. If the analyzer is automatically selecting sweep rate, you will notice the sweep slow down when the video filter is turned on.

Display Storage. Scanning spectrum analyzers typically require from a few milliseconds to several seconds to complete a sweep of the signal spectrum. At high sweep speeds, you can use an analog CRT display to view the spectrum, but at slower speeds (because of persistence – or lack of it) the trace flickers objectionably or fades out entirely before the next sweep can be started. To overcome this difficulty, early analyzers used long-persistence phosphors for *display storage*, but advances in digital technology now make it possible to digitize and store the signal spectrum electronically.

The spectrum is divided into small frequency increments, and as the analyzer sweeps through each increment the amplitude of the signal spectrum is sampled and digitized. Depending on the sweep rate, the analyzer may digitize up to 10,000 samples of the analog spectrum during each sweep. In *PEAK* display mode, the maximum values of these samples at each point during the sweep are stored in memory and displayed. A second digital display mode called MAX/MIN is also available. In this mode, the minimum sampled and digitized values from one frequency point and the maximum values from the successive point are placed in memory and displayed. The result is a display which far more closely resembles an analog display.

The digitized data are *stored* in a *waveform memory*. The stored data can then be retrieved, re-converted to an analog signal, and displayed on the screen. Even if the analyzers LO is sweeping very slowly, the trace can be refreshed from the stored data at a rate that gives it a constant and flicker-free appearance.

The contents of waveform memory are updated during each sweep. With a slow sweep speed you can actually see this happening. However, you can stop the updating process and save the stored data, making them available for later inspection or future reference. On Tektronix analyzers, you preserve the data by using the SAVE *function*. Remember: data stored in waveform memory are updated during each sweep, but saved data remain constant.

Tektronix analyzers are able to save and display more than one trace. You can use the saved waveform(s) as a reference or comparison for other measurements. The data can also be saved in *non-volatile RAM*, memory which doesn't "evaporate" when the power is turned off. This means you can make measurements in the field and then bring them back to the lab with you!

With multiple memories, you have a choice of which memory a waveform is saved in. You also have the choice of viewing or not viewing the saved waveform. If you turn off all digital waveform registers, the analog trace is displayed.

The analog display has definite advantages in two areas:

- Viewing signals which are not constant, such as video modulation or pulses. In these cases, analog display mode can be indispensable.
- High speed sweeps. The analog display can be swept many times faster than the digital display making fine-grained, rapidly changing signal characteristics visible.

The analog display feature is also convenient for comparing analog and digital spectra, providing a reassuring method of approaching digital display technology for those accustomed to earlier analog displays.

Digital storage provides additional benefits however. *Waveform subtraction* enables you to subtract a saved waveform from a stored one and display the difference. This feature is very useful when comparing the output of a device to its input, or whenever data must be normalized (e.g., when working with a tracking generator).

When two spectra are identical, their difference is exactly zero, and, therefore, any nonzero difference represents a change in the spectrum shape. This makes waveform subtraction a very sensitive detector of spectral shifts in either frequency or amplitude.

A *MAX HOLD* feature enables you to capture the maximum signal amplitude encountered during a series of sweeps. This is accomplished by comparing the digitized spectrum amplitude from the current sweep with the value stored from previous sweeps at each frequency point. If the current value is greater, it replaces the stored value and becomes the reference for comparison with the next sweep. In this way, the largest amplitudes observed over a whole series of sweeps are preserved. The MAX HOLD feature is most useful for characterizing the maximum excursions of a randomly varying signal spectrum. MAX HOLD traces can also be saved for future evaluation.

Another application of the MAX HOLD feature is to monitor a drifting oscillator or transmitter. This is accomplished by selecting a center frequency equal to the nominal oscillator frequency and enabling the MAX HOLD function. On each succeeding sweep, the frequency of the oscillator signal is measured and displayed. If the signal has drifted from its original frequency, the current sweep will have its peak amplitude at a different frequency than the previous one. That peak value is stored to memory. With repetitive sweeps, a broadened signal peak will develop. The width of this peak minus the width of the peak for a single sweep is the amount of drift in the oscillator. It is important to check the drift specifications of your analyzer to ensure the analyzer is more stable than the signal being checked.

The Tektronix 271X also has a *MIN HOLD* feature which is analogous to MAX HOLD, but preserves the minimum values. Using MAX HOLD and MIN HOLD in different waveform memories is a convenient method of defining the maximum range of signal amplitude variations.

Frequency Markers. Most spectrum analyzers enable you to place cursors, or *frequency markers*, on the stored spectrum in order to designate specific points that you want to measure. On Tektronix analyzers, the markers are intensified spots on the displayed waveform. As you will discover shortly, these markers provide the most accurate and convenient method of measuring signals. Markers are not available in analog display mode.

Center Measure and Tracking. The *CTR MEAS/TRKG control* provides the easiest method of determining signal frequency. Each time it is activated, the spectrum analyzer automatically measures the frequency of a signal peak and makes it the new center frequency (which also centers the signal). The signal that is measured will be the signal nearest center screen, or the signal nearest the marker if it is turned on. Because more than one signal peak is often present on screen, we recommend that you use the marker to designate the signal to be measured. Center measure is the most accurate and convenient method of measuring signal frequencies.

The signal track feature causes the center measure function to repeat continuously. The signal frequency is measured during each sweep and made the new center frequency. The purpose of this feature is to hold a drifting signal at center screen so that it can be observed continuously without shifting position.

Frequency Counters. Many modern spectrum analyzers are equipped with built-in frequency counters. These counters provide the ultimate in signal frequency measurement. When combined with markers and the center measure feature they enable you to determine the frequency of any signal on screen with maximum accuracy. Actual accuracy depends on the specific instrument and the input signal frequency, but the low-cost 271X series can measure a 100 MHz signal to within 60 Hertz.

Frequency Range Control. Wide range or microwave spectrum analyzers usually require several separate bands to cover their entire specified frequency range in much the same way as a communications receiver does. The *frequency range control* on these analyzers works much like the band select switch on the receiver, enabling you to select which of several bands the analyzer is currently scanning. On Tektronix 49X and 275X series instruments, each succeeding selection of the "up" or "down" control places the analyzer in a higher or lower frequency band. However, you can bypass the range controls all together by entering the center frequency from the data input keypad.

The Tektronix 271X series instruments cover their entire 1.8 GHz frequency range in one band, elim-inating the need for band selection.

Phase Lock. An analyzer usually has two or more internal oscillators. The frequency of one or more of the oscillators is changed to sweep the analyzer through a frequency band. When using wide spans, a slight amount of drift in an internal oscillator is not noticeable. However, as the span is reduced to a few kHz/div or less, any instability of the internal oscillators becomes evident. The on-screen indication is an apparently drifting signal, when the real problem is a drifting oscillator within the analyzer. Therefore, when an analyzer is operating at narrow spans, the oscillators in the spectrum analyzer are typically *phase locked* to a stable reference.

Harmonic Mixing, Preselectors, and Signal Identification. In order to analyze signals at microwave or millimeter wave frequencies, many analyzers make use of *harmonic mixing*. With this technique, the fundamental frequency of the local oscillator **and its harmonics** are deliberately mixed with the input signal. This means that anytime the difference in frequency between an input signal and the local oscillator fundamental **or any of its harmonics** equals the intermediate frequency, the signal is displayed by the analyzer. This is equivalent to having many local oscillators, each with a range of n times the fundamental (where n is the harmonic number). Thus, as the fundamental sweeps from 2 GHz to 6 GHz, the harmonics sweep from 4 GHz to 12 GHz (second), 6 GHz to 18 GHz (third), 8 GHz to 24 GHz (fourth), and so on. Up to 50 or more harmonics can be used.

The "local oscillators" in combination can therefore sweep from 2 GHz to more than 300 GHz!

This sounds great, but there is a problem – all the harmonics are present all the time. If a signal peak appears on the analyzer display, what frequency does it represent? How can you tell which harmonic it comes from? Based on what we've said so far, there's no way to tell. But we can eliminate the problem by using a pre-selector, or distinguish between frequencies by using a signal identifier.

A *preselector* is a tracking filter located ahead of the first mixer. It could be the input filter shown in Figure 5. The preselector tracks the frequency the analyzer is tuned to at any particular time and allows only a narrow band of frequencies in the range of interest to pass into the first mixer (it "preselects" the range of frequencies which will be admitted into the mixer). By switching the filter, the analyzer can be made to track with any desired harmonic. In this way, interactions between input signals and harmonics other than the desired one are prevented. An additional benefit is that any large signals present at the input but out of the frequency range being analyzed are prevented from reaching the mixer. This eliminates the need to use additional attenuation to protect the first mixer.

A preselector is generally used throughout the coax bands except for an analyzer's lowest analysis band (band one). In the lowest band, a low-pass filter serves the same purpose. Normally the pre-selector is switched automatically as you change frequency bands and you don't have to do anything. Occasionally, however, you may have to "peak" the preselector. This is usually accomplished with a front-panel control. A "peaking control" enables you to accurately center the preselector around the frequency the analyzer is tuned to (it offsets the tracking filter slightly up or down with respect to the analyzer frequency). If the filter is completely mis-peaked (offset too far from the tuned frequency), the analyzer will indicate there are no signals in the selected band. Tektronix microwave analyzers (49X, 275X, and 278X series) provide an automatic peaking mode, although manual peaking is also possible. The 271X series, being single-band analyzers, do not require a preselector.

Above about 22 GHz lie the *waveguide bands*. These are bands in which signals are typically propagated through waveguides rather than coaxial cables. Spectrum analyzers designed for use in the waveguide bands (49X, 275X, and 278X series) are equipped to use *external mixers*. Because mixing takes place outside the analyzer, preselection is not possible. Instead, waveguide band analyzers use a *signal identifier*. Signal identifiers indicate which signals are within the desired band by changing their spectrum in some way. Exactly how depends on the particular instrument. Typically, the spectrum of the signal from the desired band is shifted vertically or diagonally on alternate sweeps while the spectra of signals from undesired harmonics remain stationary.

There are two other problems associated with harmonic mixing. The first is that there is about a 3 dB loss in signal strength for each additional harmonic number (the 5th harmonic is 15 dB down relative to the fundamental). Gain is added to compensate for the loss, but the added gain raises the analyzer noise floor.

Therefore, the analyzers dynamic range decreases. The second is that local oscillator instabilities are multiplied by the number of the harmonic. For instance, if the local oscillator fundamental jitters (or FMs) by 2 kHz, the tenth harmonic will jitter by 20 kHz.

Applications

Any time an unknown signal is being measured, follow the procedures outlined in the ***Reference Level*** section of this application note to minimize chances of damaging the instrument.

Caution

In all the following applications, ensure that input signal levels do not exceed the specifications of your instrument. Exceeding specified levels can damage your analyzer.

Signal Amplitude and Frequency – The Continuous Sine Wave.

It is instructive to measure the amplitude and frequency of a sine wave signal with a spectrum analyzer because the same technique is also used to measure other types of signals. In the past, amplitude measurements were made directly from the display, and frequency measurements were made by using the analyzer to detect the zero beat between the unknown signal and a reference oscillator. Spectrum analyzers have now evolved to the point that they can directly read out signal amplitudes and frequencies with greater accuracy than the older approaches.

The steps below are a general approach to accurately measuring signal amplitudes and frequencies:

1. *Use digital display storage.* Display storage makes the use of frequency markers possible, and the analyzer internal software that displays the marker amplitude does not include display nonlinearities.
2. *Use the narrowest RBW that is wider than the signal bandwidth.* A narrow RBW offers best signal resolution, but one that is narrower than the signal bandwidth results in low amplitude readings. If the signal amplitude decreases as you narrow the RBW, you've probably made the RBW less than the signal bandwidth. Because continuous sine waves have very small bandwidths, even the narrowest RBW can be used.
3. *Adjust the reference level until the signal is within one division of the top graticule line.* The effects of log amplification are minimized when the signal is near the reference level.
4. *Place a marker on the desired signal.* The marker amplitude is displayed on screen.
5. *Use the center measure control to shift the signal to the center of the screen.* The indicated center frequency is the signal frequency.

Before measuring any signals with your spectrum analyzer, set the reference level to maximum, typically +20 or +30 dBm (+ 69 or +79 dBmV), and select maximum span. You will notice a peak in the display at zero Hertz. It is an artifact of system design that conveniently marks zero frequency. The peak is present regardless of whether or not there is an input signal. All signals to the left of the zero Hertz peak are merely images or reflections of those signals to the right of zero, not additional signal components.

Now let's measure a real signal.

1. Set the output frequency of an RF sine wave generator to about 100 MHz and its amplitude control to minimum.
2. Set the reference level and span to maximum on your analyzer, enter peak display mode, and then connect the RF generator.
3. Reduce the reference level to 0 dBm and then slowly increase the amplitude control on the signal generator until a signal is seen at the left of the screen.
4. Reduce the span of the analyzer and adjust the center frequency until the signal is approximately centered.
5. Increase the signal amplitude until the peak is within one division of the reference level.
6. Continue to decrease the span and RBW (RBW decreases automatically with span in AUTO mode) until the RBW setting is 3 kHz or less. Unless the jitter in your source is very large, the bandwidth of the sine wave will be very small, so you won't have to worry about the RBW being too narrow. Re-adjust the center frequency if necessary to keep the signal on screen.
7. Turn on a marker and place it on the signal. Press the center measure key.

The indicated center frequency is the frequency of the input sine wave; the marker amplitude readout is the amplitude of the sine wave. The marker amplitude may differ slightly from the amplitude measured with the graticule; the marker amplitude is correct because it does not include display nonlinearities. If your analyzer is equipped with a counter, the measured frequency will also be displayed on screen. See Figure 12. The "C" in front of the readouts at the upper right of the Tektronix 271X display indicates counter readings. Compare this measured result with Figure 2B.

A Square Wave Spectrum.
Use this procedure to view the spectrum of a square wave:

1. Set the reference level of your spectrum analyzer to maximum, the center frequency to 500 kHz, and the span to 100 kHz/div.
2. Set the output frequency of a square wave generator to 100 kHz and its amplitude control to minimum; connect the generator to the analyzer.
3. Reduce the reference level to 10 dBm and increase the signal generator amplitude control until signal peaks are seen on the analyzer. Continue increasing the amplitude until the peak at 100 kHz is above mid-screen.

Figure 13 shows how the signal should appear. Compare this measured result with Figure 4B. The analyzer displays a "fundamental" sine wave at the same frequency as the square wave (100 kHz) and other components of diminishing amplitude at multiples of

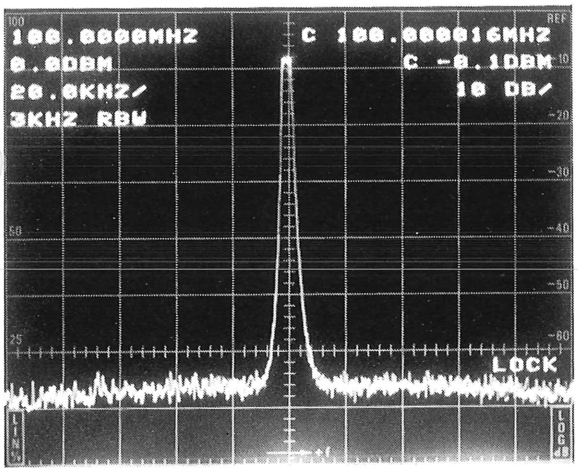


Figure 12. Spectrum of a simple sine wave.

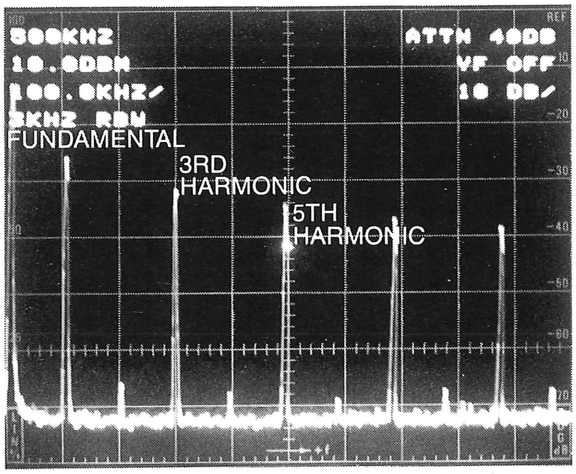


Figure 13. 100 kHz square wave spectrum.

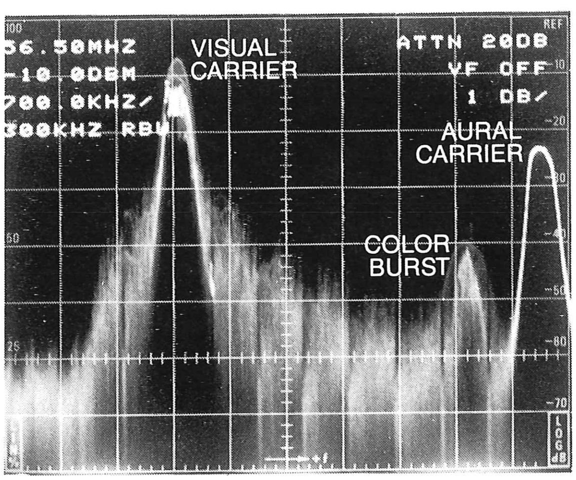


Figure 14. Broadcast TV signal spectrum.

the fundamental. These other frequencies are identified as the 3rd, 5th, 7th, etc. (odd) harmonics of the fundamental frequency. If the even harmonics are present, the duty cycle of the square wave is not exactly 50%. This is a very sensitive test of the "squareness" of the square wave. You can see low level even harmonics in Figure 13 indicating that the signal is not quite square.

A Broadcast Signal Spectrum. With most analyzers, you can directly view broadcast radio and television signals. Here's how:

1. With the reference level and span at maximum, connect a TV antenna (often just a short wire antenna will work) to the input of your analyzer. Momentarily ground the lead first to be certain there is no large static charge present.
2. After ensuring no unduly large signals are present, reduce the span to about 1 MHz/div, and set the center frequency near the video carrier frequency of a local TV station (try the 50 - 70 MHz range if you are unsure of the exact frequency).
3. Adjust the reference level until the spectrum of the TV signal is above half-screen, and turn off all waveform registers. Compare your display to Figure 14, which demonstrates the signal detail that can he seen in analog display mode.

Modulation

In communications, information is transmitted as changes in a signal. Changing some characteristic of a signal as a function of time is called *modulating* the signal. How can a signal be changed? Two common methods are to vary the signal's amplitude or to change its frequency. The process of changing the signal and the resulting alterations to the original signal are both called *modulation.* If we change the amplitude, it is called *amplitude modulation (AM)*, and if we change the frequency, it is called *frequency modulation (FM)*.

Amplitude Modulation. In amplitude modulation, the amplitude of a sine wave signal is varied in accordance with the amplitude of another signal. We can modulate the original sine wave by multiplying it by a constant plus a second, lower frequency sine wave. The lower frequency sine wave is referred to as the

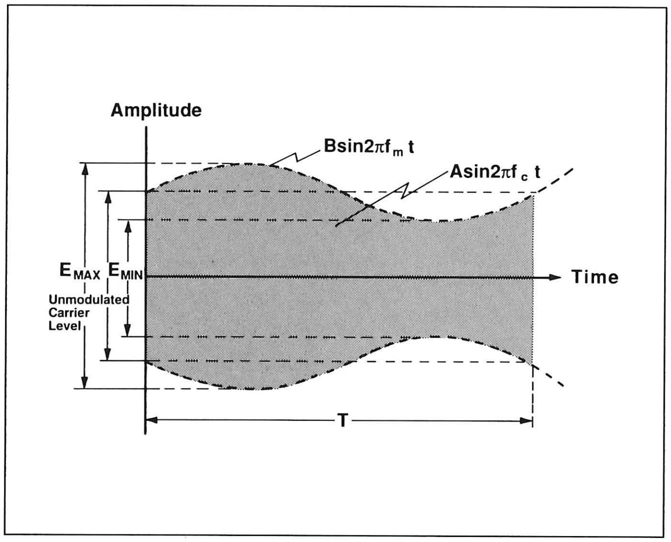


Figure 15A. Time domain drawing of sine wave amplitude modulation.

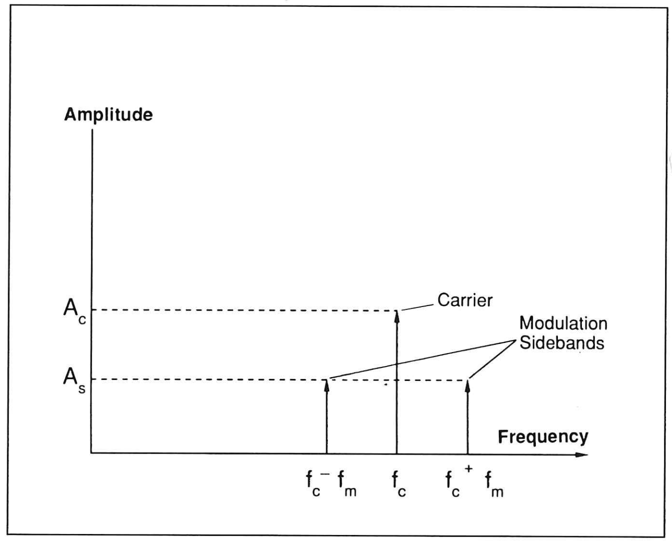


Figure 15B. Theoretical frequency domain representation of the sine wave amplitude modulation in Figure 6A.

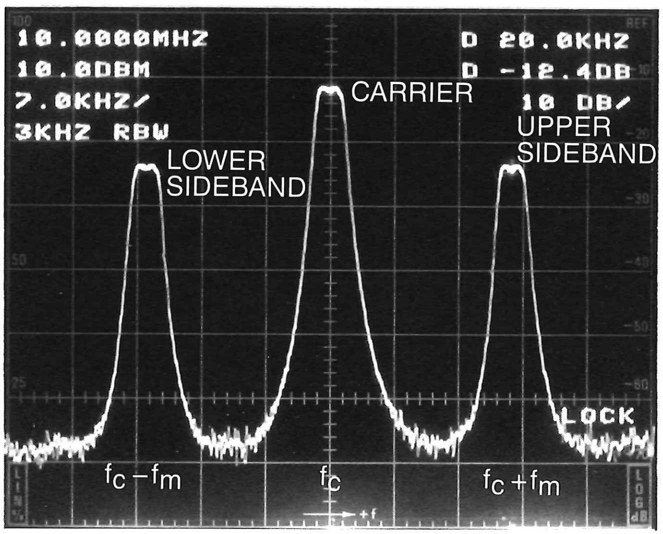


Figure 16. Amplitude modulation in the frequency domain.

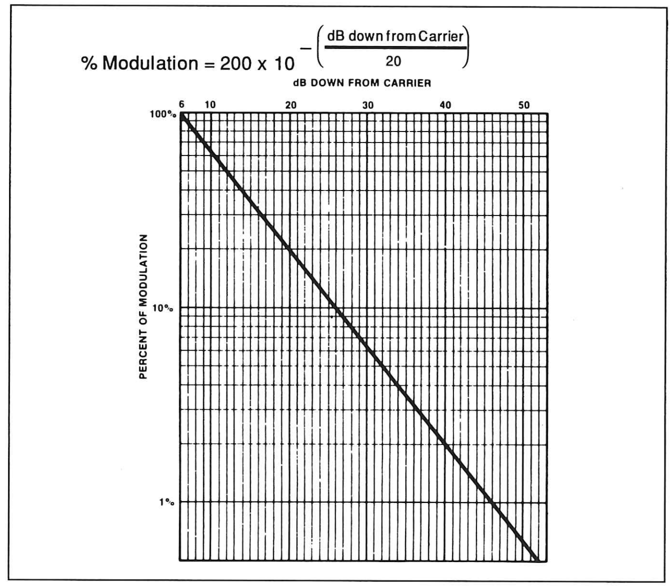


Figure 17. Chart of sideband level vs. % modulation.

modulating signal. This approach to modulation makes the amplitude of the original sine wave vary about a constant value at the frequency of the lower frequency sine wave. Figure 15A shows how this signal appears in the time domain. The slowly varying sinusoids which outline the amplitude of the faster sine wave are called the *modulation envelope* (or sometimes simply "the modulation") and have the same frequency as the modulating signal. The faster sine wave, which has the same frequency as the original sine wave, is called the *carrier signal,* or simply the *carrier.* The *modulation factor* is a measure of how much the modulation varies the amplitude of the carrier; it's equal to the ratio of one half the peak-to-peak variation of the modulation envelope to the peak-to-peak amplitude of the unmodulated carrier. *Percent modulation* equals 100 times the modulation factor. The signal is 100% amplitude modulated when the maximum peak-to-peak amplitude of the modulated signal equals twice the peak-to-peak amplitude of the unmodulated carrier, and the minimum peak-to-peak amplitude of the signal is zero. Percent modulation is often measured in the time domain as shown in Figure 15A, but in the case of sine wave modulation, it can be easily measured in the frequency domain with a spectrum analyzer. We'll show you how shortly.

Figure 15B shows the frequency domain representation of the same amplitude modulated signal. The spectrum consists of three sine wave components. The central signal, which represents the carrier, occurs at the frequency of the unmodulated sine wave (*the carrier frequency*) and its amplitude is A_C. A pair of sine waves, called *modulation sidebands,* occur at frequen-

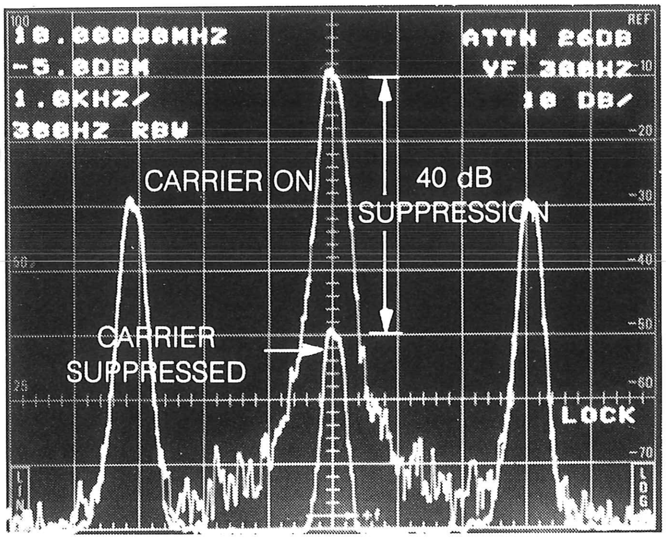


Figure 18. Suppressed carrier modulation.

cies equal to the sum and difference of the carrier frequency and the modulating signal frequency, $f_c + f_m$ and $f_c - f_m$. The amplitude of each sideband is A_s. There is a pleasing symmetry. In the time domain the carrier was bounded above and below by equal amplitude sine waves at the modulating signal frequency, whereas in the frequency domain the carrier is surrounded by equal amplitude sine waves which differ from it by the modulating signal frequency.

Figure 16 shows how a real AM signal appears on a spectrum analyzer. Compare Figure 16 with Figure 15B. Notice that in Figure 16 the carrier frequency (which equals the center frequency) is 10 MHz. The sidebands are separated from the carrier by about 2.8 divisions, indicating a modulation frequency of 20 kHz. Notice that there is noise present in the "real" spectrum, and that the signal peaks are somewhat broadened, due primarily to the width of the RBW filter. You can determine the frequency of the carrier and the frequency of the modulation even more accurately by using the marker method described earlier in the ***Signal Amplitude and Frequency*** section.

The percent modulation can be found by noting the difference between the carrier and sideband amplitudes. If your analyzer has a delta marker mode, you can use it to conveniently measure the carrier-to-sideband difference:

1. Place a single marker atop one of the sidebands and operate the center measure control. The sideband is now centered.
2. Enter delta marker mode and place the second marker atop the carrier; operate the center measure control.

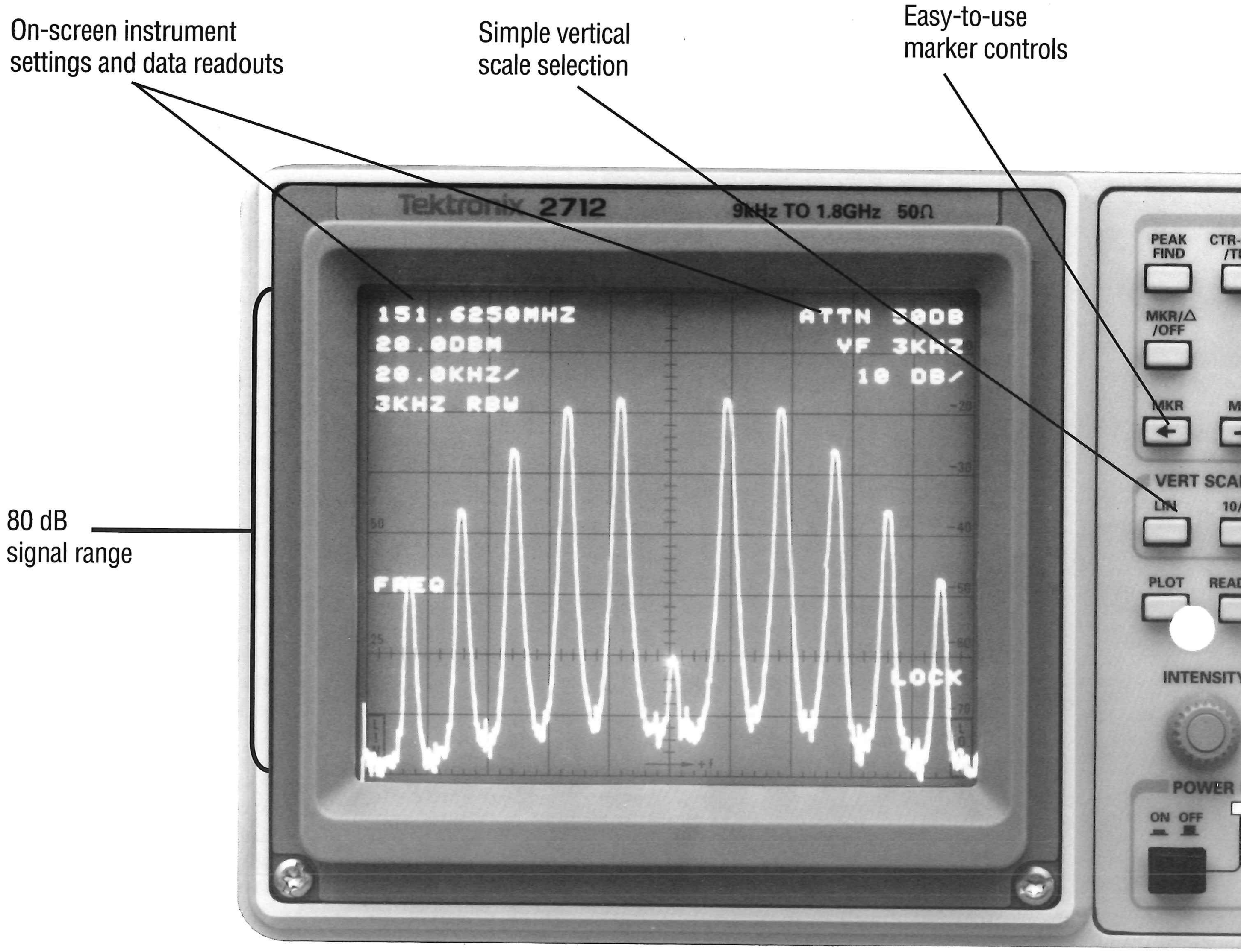
The spectrum is shifted so that the frequency of the carrier becomes the new center frequency. The differences in amplitude and frequency between the carrier and the sideband are read out on screen with a "D" ("DC" when the counter is installed) in front of them as in Figure 16. From the figure, the sidebands are approximately 12 dB below the carrier. Use the difference to read the percent modulation from the chart in Figure 17. For a 12 dB difference, the chart indicates the percent modulation is 50%.

The spectrum of a *suppressed carrier* signal can also be displayed with a spectrum analyzer. The spectrum would appear as in Figure 18, which is a composite photograph showing the carrier present and suppressed. Carrier suppression is the difference in carrier amplitude between the unsuppressed carrier and the carrier after suppression. Figure 18 indicates the carrier is suppressed by 40 dB.

A similar approach can be used in the case of *sideband suppression*. You can determine the amount of suppression by measuring the difference in the suppressed sideband's amplitude before and after suppression. If you assume the sidebands are the same amplitude before suppression, you can approximate the amount of suppression by measuring the difference in amplitude between the upper and lower sidebands.

Flatness of Audio Pass Band. Another measurement that can be made on an AM system is to determine its audio response. A low frequency spectrum analyzer or a high performance RF spectrum analyzer would be needed to perform this measurement at base band. Base band is the lowest frequency band in which a signal normally exists; in this case, the audio band of the modulation system. However, relatively low cost heterodyne analyzers can be used to make the measurement at RF frequencies.

Slowly sweep the system's input with an audio generator of known flatness. Monitor the resulting RF signal with a narrow span/div and a vertical scale factor of 1 or 2 dB/div (you may have to drive the carrier peak off-scale to do this). By using the MAX HOLD function, you can construct a trace showing the shape of the audio pass band. Figure 19 is a photo of such a sweep. The shape of the spectrum also indicates any emphasis, or spectrum shaping, placed on the audio.

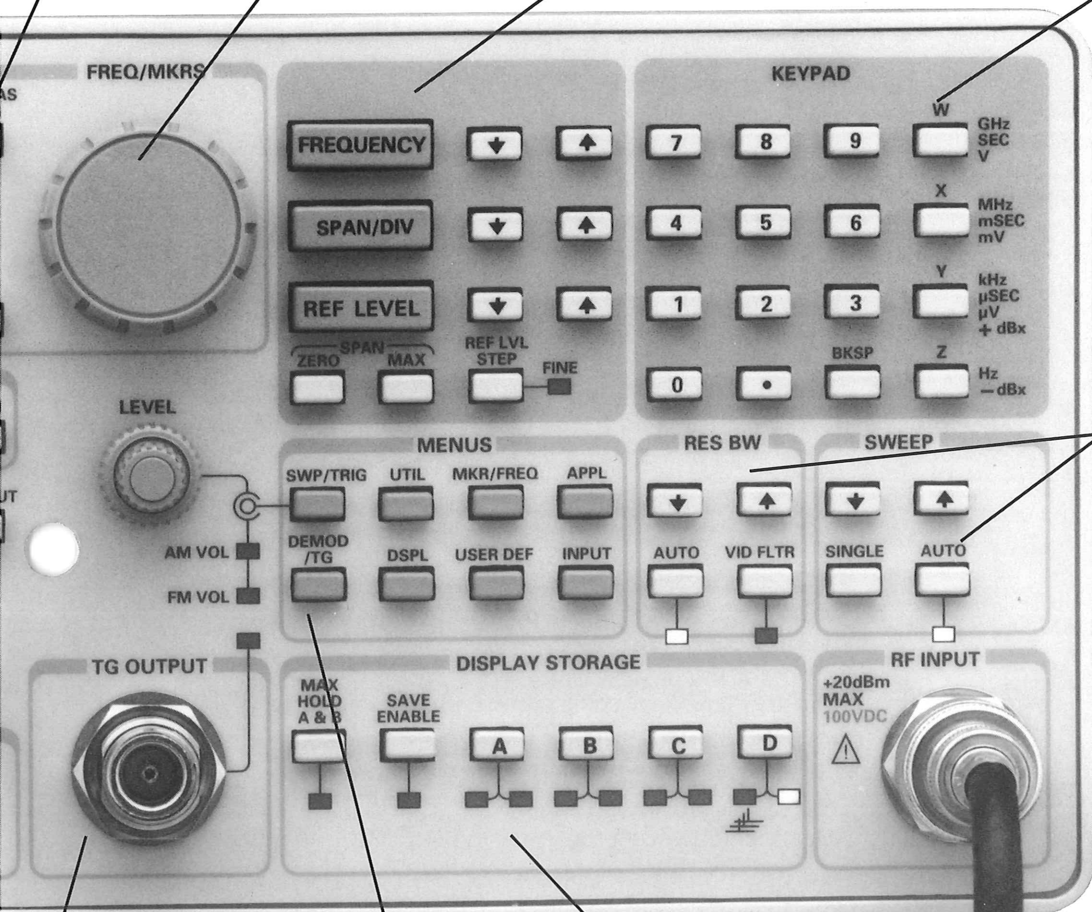

On-screen instrument settings and data readouts
Simple vertical scale selection
Easy-to-use marker controls
80 dB signal range
Tektronix 2712
9kHz TO 1.8GHz 50Ω
151.6250MHZ
20.0DBM
20.0KHZ/
3KHZ RBW
ATTN 50DB
VF 3KHZ
10 DB/
FREQ
LOCK
PEAK FIND
MKR/Δ /OFF
VERT SCAL
PLOT
INTENSITY
POWER
ON OFF

Standard center measure, signal track and counter functions

Manual frequency/ marker control with "constant rate" tuning

Direct numerical entry or step-up, step-down control of primary measurement parameters

Keypad for numerical entry and menu selection



Automatic or manual control of sweep rate and resolution bandwidth

Optional built-in tracking generator

Convenient display storage functions – up to four waveforms simultaneously

Menu keys co-located for ease of selection

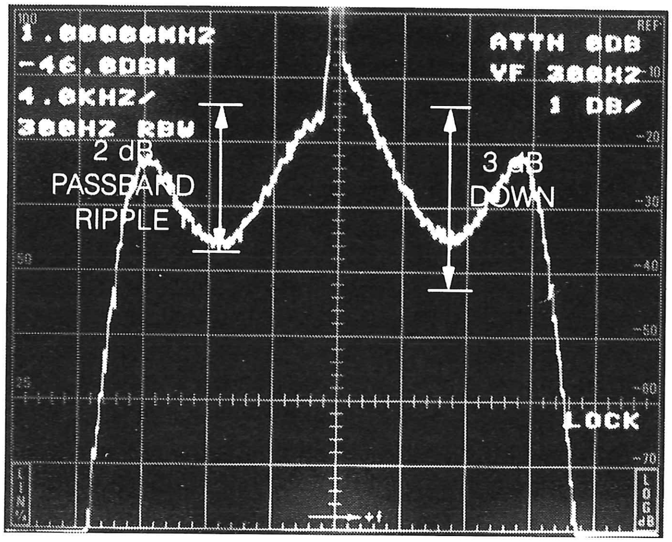


Figure 19. Audio response of AM signal at carrier frequency.

From Figure 19 we can see that the passband ripple is about 2 dB and that the 3 dB audio bandwidth is in excess of 12 kHz. The lower and upper sidebands should be symmetrical. If the transmitter were seriously mistuned or working into a poor antenna match, the spectrum analyzer would show how each sideband is individually affected.

Distortion. *Distortion* is the result of electronic circuits operating in a non-linear mode. The two most common types of distortion are known as harmonic distortion and intermodulation distortion. Typically you measure distortion by driving the equipment inputs with known signals and monitoring the equipment output for signal components other than those present at the input. When making distortion measurements, it is important to be sure the modulating signal itself is free from any distortion products. If in doubt, check the signal source at baseband with a spectrum analyzer.

Harmonic distortion is the distortion that results when a signal interacts with itself due to non-linearities in the equipment to produce sidebands at multiples, or harmonics, of the frequency components of the original signal. For instance, problems within a modulator or audio amplifier stage, or amplitude modulation in excess of 100%, can result in harmonics of the modulating signal. On the other hand, overdriving an RF oscillator or amplifier can result in the generation of harmonics of the carrier signal.

As an example of a typical harmonic distortion measurement, you might drive the audio input of a modulator with a variable frequency oscillator set at a specified frequency. The modulator is driven to a specified percent modulation and a spectrum analyzer is used to check the output for the presence of any signals other than the carrier and the modulating signal sidebands. If harmonic distortion occurs, additional sidebands can appear on the screen at multiples of the carrier or modulating frequency. If the modulator is driven with a 5 kHz test signal, as in Figure 20, harmonic distortion of the modulating signal shows up as sidebands at 10 kHz, 15 kHz, 20 kHz, etc, from the carrier.

The percent harmonic distortion of any sideband can be found by noting the difference in dB between the fundamental and the harmonic, and determining the percentage from the chart in Figure 21. Using the values from Figure 20, the percent distortion of each harmonic is:

2nd harmonic = 3.25%
3rd harmonic = 1.2%
4th harmonic = 0.18%

The *total harmonic distortion (THD)*, or the percent distortion due to all sidebands, can be found like this:

1. Convert the percent distortion due to each harmonic to a decimal fraction
2. Square the decimal fraction of each harmonic
3. Add the squared values
4. Take square root of the sum
5. Multiply by 100

This procedure is accurate only if the upper and lower sidebands of each harmonic pair are within one or two dB of each other.

For the values from Figure 20, the THD is given below.

THD of Figure 20
$= (0.0325^2+0.012^2+0.0018^2)^{1/2}$
$= 0.035 = 3.5\%$

Intermodulation distortion is the distortion that results when two or more components of the input signals interact to produce sidebands at frequencies equal to the sums and differences of the original components. The *two tone intermodulation distortion (IM) test* is a common intermodulation measurement. In this test, the outputs of two audio generators are combined, and the result is applied to the input of the device under test (DUT). The method of combining the two signals is very important, as mixing the two sources can itself create unwanted products. Combining should occur in a directional Bridge. A "T" connection or combiner can be used, provided each generator is sufficiently padded. Use a

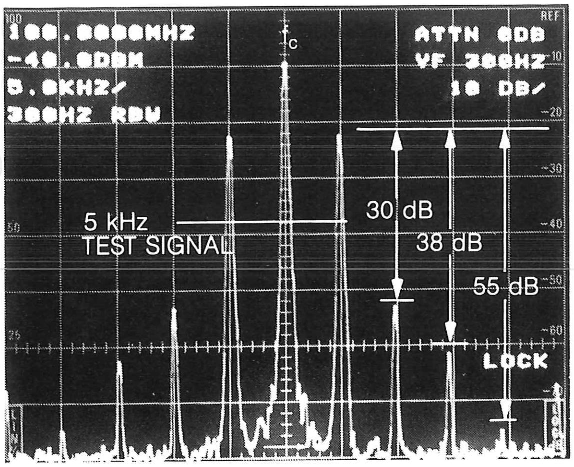


Figure 20. Spectrum showing harmonic distortion.

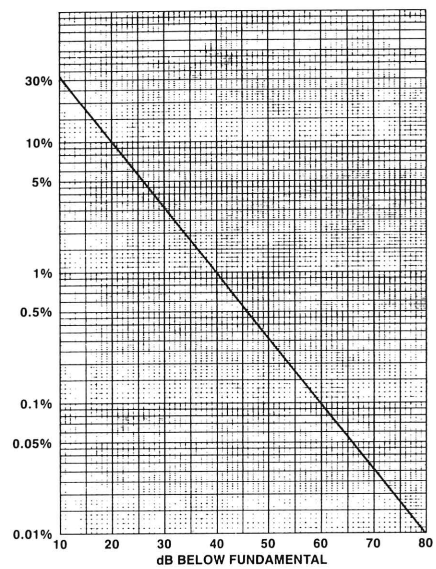


Figure 21. Chart of % harmonic distortion as function of difference in sideband levels.

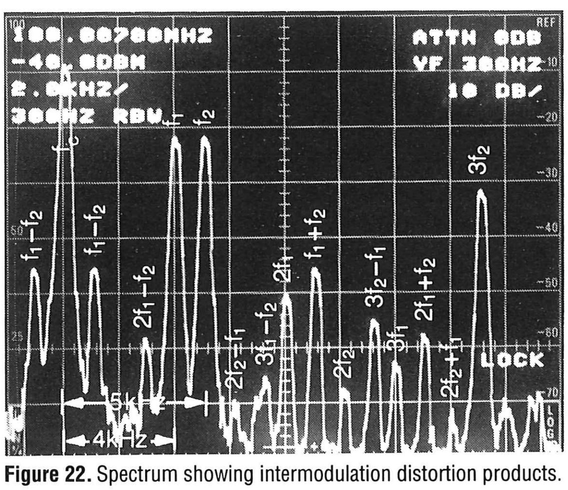


Figure 22. Spectrum showing intermodulation distortion products.

spectrum analyzer to check the output of the directional bridge or combiner to ensure no signals other than the desired two tones are present. The frequency of the input signals used depends on the type of test to be performed and the type of equipment being checked. After the combined signals are injected into the DUT, monitor the output of the device with a spectrum analyzer for intermodulation distortion products. If a transmitter output is used as the monitor point, be certain there is sufficient external attenuation to protect the analyzer.

A modulator output, with intermodulation distortion present, is shown in Figure 22. In this example, 4 kHz (f_1) and 5 kHz (f_2) modulating signals are used. Many IM products are created; the first set, or second order IM products, are separated from the carrier by $f_1 + f_2$, $f_1 - f_2$ and/or $f_2 - f_1$ (9 kHz and 1 kHz from the carrier). The third order IM products occur at $2f_1 + f_2$, $2f_1 - f_2$, $2f_2 + f_1$, and/or $2f_2 - f_1$ (13 kHz, 3 kHz, 14 kHz, and 6 kHz). Higher order products are also created, but their amplitude generally decreases with order number.

Frequency Modulation. Frequency modulation is a bit more complicated than amplitude modulation. Here the amplitude of the carrier is held constant but its frequency is varied in accordance with the amplitude of the modulating signal. The frequency of the modulated signal at any instant in time is called the *instantaneous frequency*. The instantaneous frequency is equal to a constant carrier frequency, f_c, plus or minus an amount that depends on the amplitude of the modulating signal. That is, the value of the instantaneous frequency depends only on the amplitude of the modulating signal and not its frequency. However, the modulating signal frequency does determine how fast the instantaneous frequency changes. The maximum difference between the instantaneous frequency and the carrier frequency is called the *frequency deviation*, ΔF, and is usually set by the hardware used for the modulation process. It expresses the frequency difference between the carrier and the instantaneous frequency when the modulating signal is at maximum specified amplitude. Therefore, if we modulate an FM carrier with a sine wave of maximum specified amplitude, we should get a signal which is constant in amplitude but whose instantaneous frequency varies between $f_c + \Delta F$ and $f_c - \Delta F$ at the frequency of the modulating sine wave. In U.S. broadcast FM, for instance, 100% frequency deviation is 75 kHz[1].

[1] When SCAs are present, the FCC permits this to be increased to 110%.

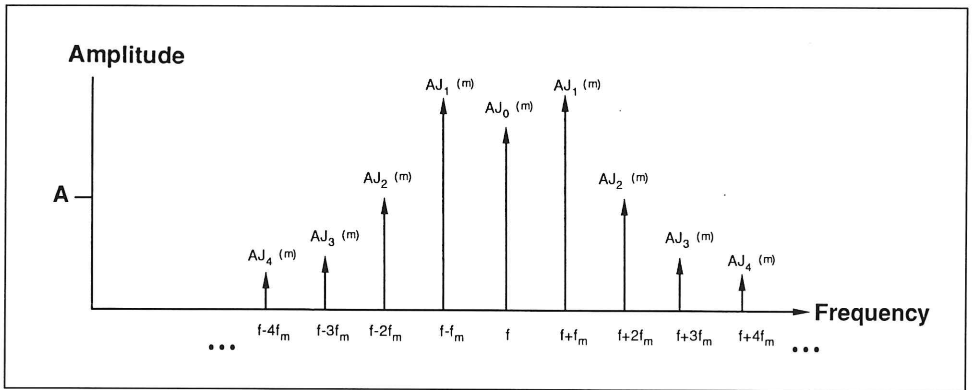


Figure 23. Theoretical frequency domain representation of sine wave frequency modulation.

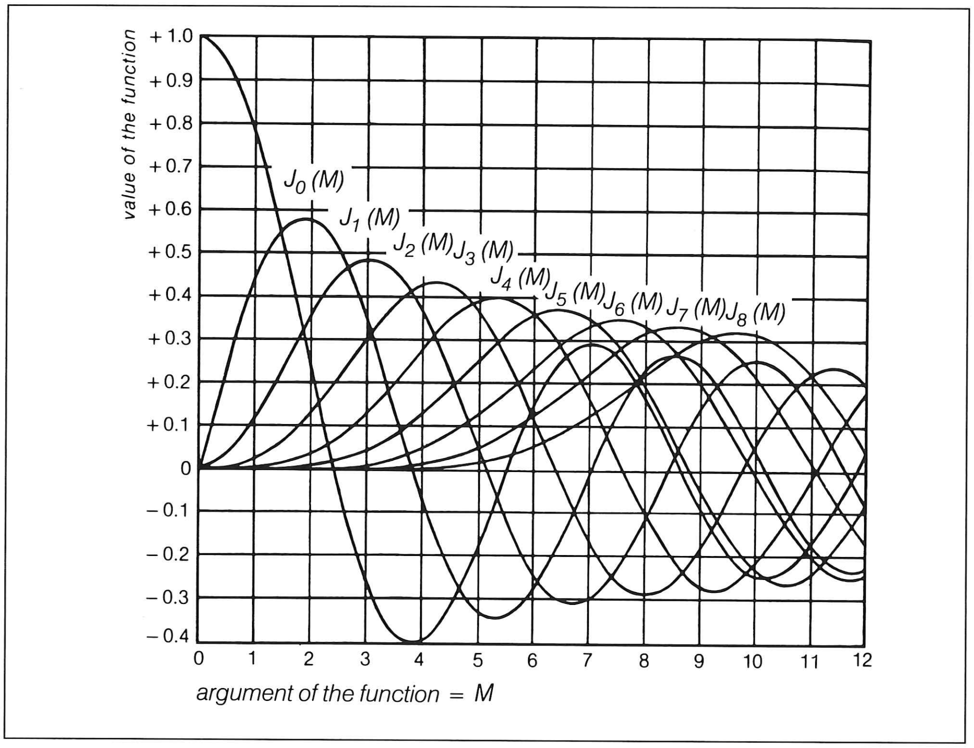


Figure 24. The first eight orders of Bessel functions.

What does this signal look like in the frequency domain? Theoretically, for a single modulating tone, the spectrum of the FM signal consists of a carrier frequency component and an infinite number of sidebands spaced at harmonics of the modulating frequency. This is shown in Figure 23 for a single modulating tone at frequency f_m. The amplitudes of the carrier frequency and the sidebands are proportional to (get ready for this!) Bessel functions evaluated at the modulation index.

What is a Bessel function? Or a modulation index? Neither is all that mysterious. A *Bessel function* is a solution to a particular type of equation just as a sine wave is the solution to another type. The *modulation index*, M, is the frequency deviation divided by the frequency of the modulating signal. Therefore, the modulation index decreases as the modulating frequency increases for signals with the same amplitudes. The first eight Bessel functions are shown in Figure 24. When calculating FM spectra, they are usually written as $J_0(M)$, $J_1(M)$, and so on. The subscript indicates the *order* of the Bessel function and is the same value as the order of the sideband; M is the modulation index. If the amplitude of the original FM signal is A, then the amplitude of the carrier is $AJ_0(M)$; the first order sidebands are $AJ_1(M)$; the second order sidebands $AJ_2(M)$, and so on.

What is equally important is that for certain values of M, the Bessel functions have a value of zero. That is, for certain modulation indices, the carrier or any of the sidebands can be made to disappear. This can be used as an accurate measure of frequency deviation.

You can view the spectrum of a FM transmitter or device by connecting a spectrum analyzer to an RF test point at the output of the transmitter or device, or to a suitable antenna. Be certain to observe all cautions for safeguarding the analyzer input stages, especially when monitoring a test point at the transmitter output. Signal levels, frequencies, and other parameters can be measured in the case of FM signals just as in the case of AM signals.

Let's use an FM stereo broadcast station as an example:

1. Adjust the spectrum analyzer center frequency to the carrier frequency of the station, set the span to 20 kHz/div, and the RBW to 3 kHz.
2. Attach a suitable antenna (or CATV tap) to the analyzer input.
3. Adjust the reference level so that the spectrum can be clearly seen.

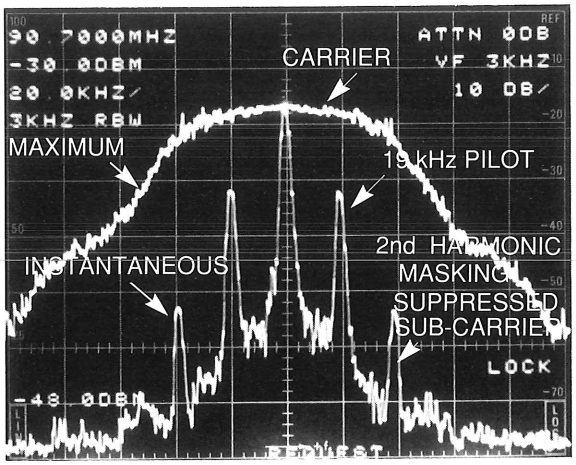


Figure 25. Instaneous and maximum spectra of an FM broadcast signal.

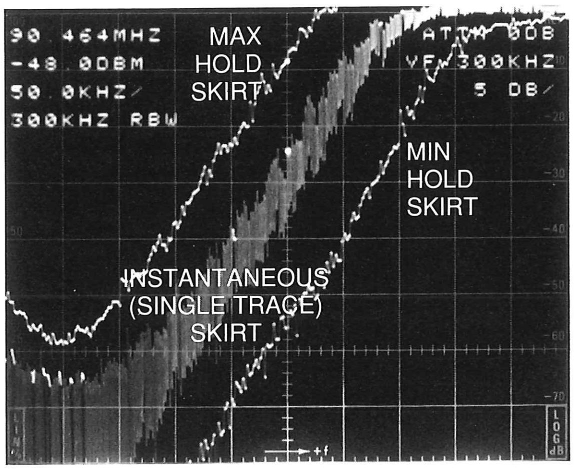


Figure 26. Determining deviation by measuring skirt width.

You can quickly spot any spurious out-of-band signals. If you wait until a period when there is no significant audio modulation, you should see a spectrum similar to the lower trace in Figure 25. The 19 kHz pilot tone is visible 16 dB below the carrier, and what appears to be the suppressed stereo sub-carrier is visible 38 kHz from the carrier[2]. There is no subsidiary communications authorization (SCA) on this station.

You can also view the spectrum occupied by the signal by setting up your analyzer as above and then activating the MAX HOLD feature. As the broadcast signal varies through its normal range of values, a broad-humped spectrum similar to the upper trace in Figure 25 will develop. The total width of this spectrum is considerably more than the 150 kHz you might expect based on a ±75 kHz ΔF. This is because each component of the modulating signal generates a theoretically infinite series of sidebands (see Figure 23) which, observed over a period of time, produce the spectrum that you see. It also helps explain why, in the US, the FCC channel assignments for FM stations are not closer than 800 kHz in the same geographical area.

Direct Verification of Frequency Deviation. One approach to measuring frequency deviation is to observe the skirts of the spectrum while using an RBW filter that is broader than the signal bandwidth. Imagine that the signal consists of a single sinewave of frequency f_s. The spectrum corresponding to this signal is just the RBW filter shape centered at f_s. Slowly vary the frequency between $f_c - \Delta F$ and $f_c + \Delta F$. As the signal frequency varies between $f_c - \Delta F$ and $f_c + \Delta F$, the shape of the RBW filter is replicated at each frequency point. The resulting spectrum, seen over a period of time, will be the superposition of all the RBW filter shapes, and the width of its skirt will be $(f_c + \Delta F) - (f_c - \Delta F) = 2\Delta F$. Because any signal can be represented as the sum of a series of sinusoids, this technique can also be used to view the deviation of a modulated signal. Use MAX HOLD in one register and MIN HOLD in another to capture the extremes of the deviation. Follow this procedure:

1. Connect the analyzer to the selected test point, and implement these settings:
   a. Center Frequency: carrier frequency
   b. Span: 50 kHz/div
   c. RBW: 300 kHz
   d. Display Mode: MAX/MIN
2. Adjust the reference level so the signal peak is near the top of the screen.
3. Center the left-hand skirt.
4. Increase signal height until the 5 dB or 1 dB per division scale can be used. You can also change the span, if desired, to increase resolution.
5. Activate MAX HOLD in register A and MIN HOLD in register C. Allow several minutes for the display to build up. The display should resemble Figure 26. The center trace is the result for a single sweep.
6. Measure the width between the MAX and MIN traces along any horizontal line. You can do this by counting graticule divisions between the traces and multiplying by the scale factor. You can also use a single marker to measure the frequency first at the intersection of the MIN trace and a horizontal graticule line, then after turning off the C register, at the intersection of the MAX trace and the same graticule line. Subtract the two readings to find the width of the skirt. This is the peak-to-peak frequency deviation. Accuracy is in the 5 - 10% range.

[2] It is the second harmonic of the 19 kHz pilot that you actually see; it is larger in amplitude than the suppressed subcarrier, and masks it from view. To see the true subcarrier, it is necessary to turn off the pilot.

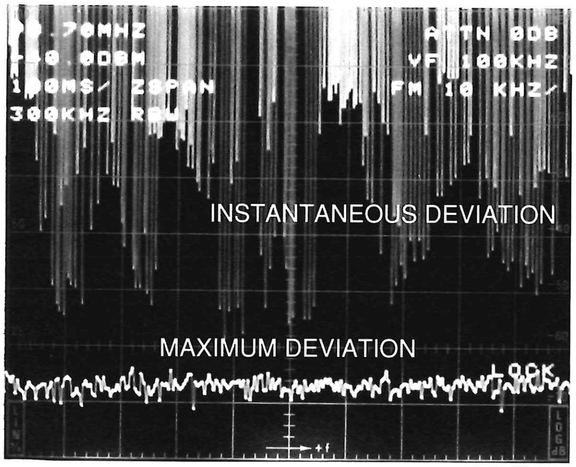


Figure 27. Determining deviation using the FM Deviation/FM Detector mode.

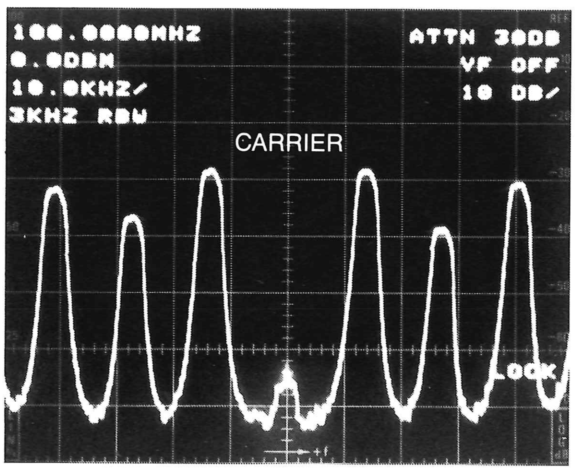


Figure 28. Typical spectrum showing second Bessel carrier null.

Table 1. Bessel function nulls.

		Modulating Signal Sidebands		
	Carrier Signal	First	Second	Third
Null No.	$J_0(M)$	$J_1(M)$	$J_2(M)$	$J_3(M)$
1	2.4048	3.8317	5.1356	6.3802
2	5.5201	7.0156	8.4172	9.7610
3	8.6531	10.1735	11.6198	13.0152
4	11.7915	13.3237	14.7960	16.2235
	Modulation indices at which Bessel functions are zero			

The Tektronix 271X Spectrum Analyzers also provide a special mode of operation to directly monitor the instantaneous frequency deviation. Use this procedure:

1. Tune to an FM carrier as above. Ensure the RBW is set to either 300 kHz or 500 kHz.
2. Adjust the reference level until the signal is above mid-screen.
3. On the 2710 press [DSPL MENU]/7/2 to select the FM Detector.

   On the 2712 press [APPL]/[7] to select the FM Deviation Mode. A display similar to the upper trace in Figure 27 will appear. It represents the instantaneous deviation of the signal frequency below the carrier (the upward deviation is not viewable). The vertical scale has become a very fast frequency meter. Zero deviation is the top graticule line and each vertical division represents 10 kHz (but you can change the scale to 5 kHz/div or 1 kHz/div using the 10 dB/5 dB/1 dB key if desired). In other words, a signal which descends seven divisions below the reference line has deviated 70 kHz from the carrier.
4. Activate MIN HOLD in register C, and allow several minutes for the display to develop. A ragged line similar to the lower trace in Figure 27 will develop. It represents the maximum downward deviation during the observation period. Its peak values should approach 75 kHz for US broadcast FM stations without SCAs.

Frequency Deviation Using the Bessel Null Method. The most accurate way to determine frequency deviation is to use the *Bessel null method.* The Bessel null method relies on the fact that the Bessel functions are zero for certain values of the modulation index (see Figure 28). Because the method cannot be used in an on-air broadcasting situation, it is usually used off-air to calibrate the station's modulation meter.

Table 1 lists the values of the modulation index for which several of the Bessel functions are zero. The FM carrier is proportional to $J_0(M)$ and each modulation sideband is proportional to $J_n(M)$, where n is the sideband number and M is the modulation index. The modulation index is just the peak frequency deviation divided by the modulating frequency. Therefore, we can make the carrier or a sideband disappear from the FM signal spectrum by choosing a modulating frequency that corresponds to one of the modulation indices in Table 1.

To establish the 100% modulation point for a particular deviation frequency, follow this procedure:

1. Decide which signal component you will null. To calculate the required frequency, divide the deviation frequency by the corresponding modulation index from Table 1. For instance, let's suppose the deviation is ±75 kHz. Then the first two carrier null frequencies are 31.188 kHz (75 kHz/2.4048) and 13.586 kHz (75 kHz/5.5201).
2. The audio pass band of most stations will not transmit 31 kHz, so select the second null value of 13.586 kHz.
3. Connect an audio sinewave generator to the modulator input and set its frequency to about 13 kHz.
4. Set the amplitude of the audio signal to zero. Connect your spectrum analyzer to an RF test point or directional coupler at the transmitter output.
5. Increase the audio generator output level until modulation sidebands are visible. Use the 271X delta-counter mode to set the difference frequency between the carrier and the first sideband to 13.586 kHz (alternately, you can use a frequency counter to measure the frequency at the audio generator output). The accuracy of the Bessel null method is governed by the accuracy of the modulating signal.

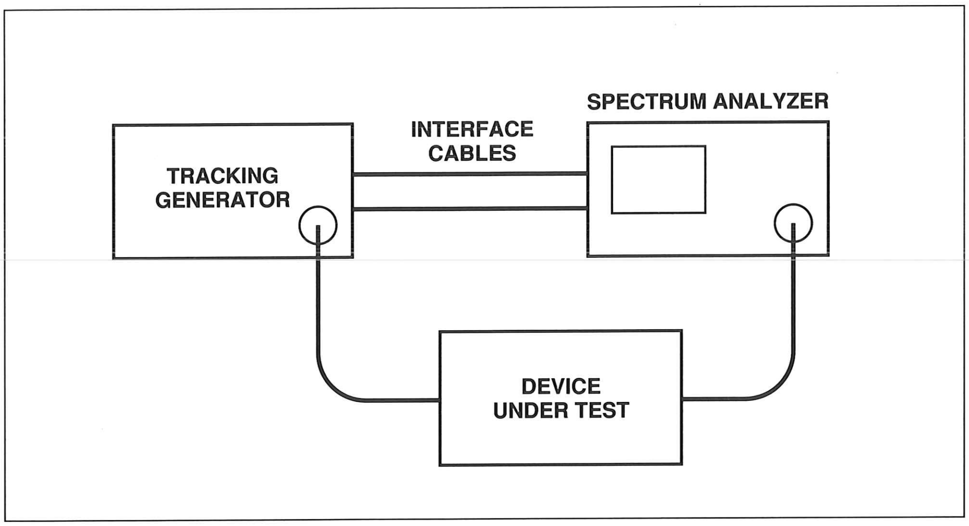


Figure 29. Tracking generator/spectrum analyzer test setup.

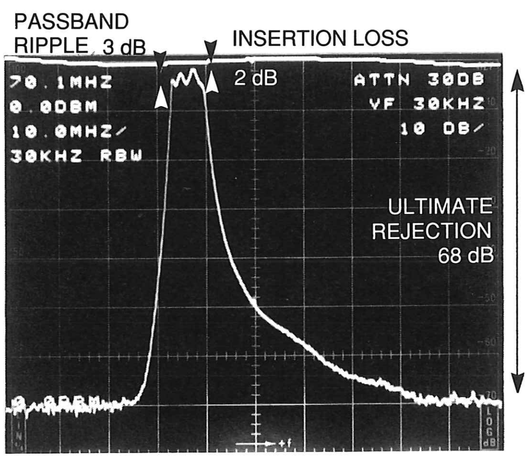


Figure 30. Filter response using tracking generator/spectrum analyzer combination.

6. If your deviation meter is already approximately correct, increase the signal generator output level until the meter reads about 100% and then vary the signal level slightly until the analyzer shows a null.
7. If your meter is totally uncalibrated, increase the signal generator output level until a null is reached. Continue increasing the output level until the second null occurs.

A representative display of the spectrum near the null-carrier condition is shown in Figure 28. The carrier null represents exactly 100% modulation. The modulation meter should now be adjusted to show 100%.

The Bessel null method can be used for any percent deviation by multiplying the frequency required for 100% deviation by the alternate percentage. For instance, in the example above, the second carrier null using a frequency of 6.793 kHz (half of 13.586 kHz) represents 50% deviation.

Tracking Generator – with Spectrum Analyzer for Swept Measurements

A *tracking generator* (TG) is a signal generator whose output frequency is synchronized to, or tracks with (hence "tracking generator"), the frequency being analyzed by the spectrum analyzer at any point in time. When used with a spectrum analyzer, a tracking generator allows the frequency response of filters, amplifiers, couplers, etc. to be measured over a very wide dynamic range. Furthermore, because the signal being analyzed is always centered in the RBW filter, the resulting display is essentially independent of the filter shape or bandwidth. Measurements are performed by connecting the output of the tracking generator to the input of the device being tested, and monitoring the output of the device with the spectrum analyzer. Figure 29 shows how the equipment is set up. The spectrum analyzer and tracking generator must, of course, be designed to work with each other.

The response displayed on the screen of the analyzer is the combined "unflatness" of the tracking generator/spectrum analyzer system and the response of the device being tested. The unflatness of the measurement system can be removed by using the B-SAVE A or B,C MINUS A function of the analyzer. First, connect the tracking generator directly to the spectrum analyzer and save the flatness (or unflatness) of the combination in the A memory by using the SAVE function. Be sure to use the same vertical display mode that will be used to make the measurement. Then, connect the tracking generator to the device being tested and monitor the device with the analyzer. The analyzer display indicates the response of the entire system (tracking generator + device + analyzer). By activating the B-SAVE A mode, the saved unflatness of the measurement system is subtracted from the response of the device being tested plus the measurement system, and the corrected display shows only the frequency response of the device being tested.

Filter. Figure 30 shows a 57 MHz bandpass filter being swept by the tracking generator/analyzer system. The *filter loss* is approximately 2 dB as noted from the difference in amplitude between the tracking generator/spectrum analyzer response (upper trace) and the filter response (lower trace). There is about 3 dB ripple in the pass band, which is about 6 MHz wide. The filter *ultimate rejection* is about 68 dB.

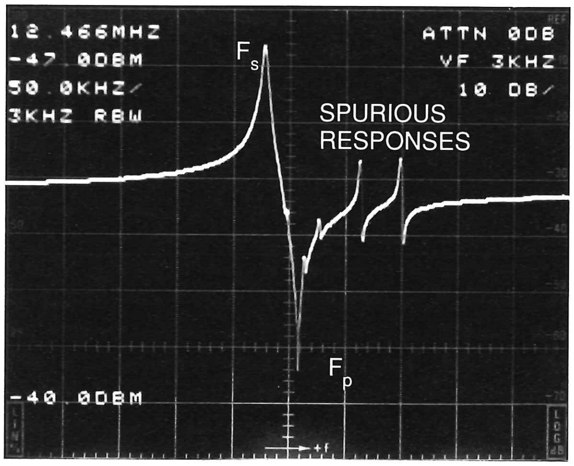


Figure 31. Crystal response using tracking generator/spectrum analyzer combination.

Crystal. Figure 31 shows a typical frequency response for a 12.5 MHz crystal. The *series resonant frequency* (f_s) and *parallel resonant frequency* (f_p) are identified on the photo. Also notice the crystal-generated spurious responses to the right of the series resonance. The TG/analyzer reference trace is not shown, because in the case of crystals, you are usually more interested in the resonances than the insertion loss. However, it could be displayed, or even subtracted out using the B,C MINUS A feature.

How the crystal is hooked up can be very important. For more information be sure to read ***Crystal Device Measurements Using the Spectrum Analyzer***, Tektronix application note 26AX-3525.

Amplifier. In Figure 32, a cable TV amplifier is being tested. Minimum loss pads have been used on the TG output and analyzer input to match to the amplifier 75 ohm impedance. External attenuation values have been entered (–4.0 dB for the TG pad DEMOD/TG/8/1 on the 2712; –7.5 for the analyzer pad using INPUT MENU/6/1). Hence, the reference level and tracking generator readouts indicate "OFST". Note that the vertical scale has been switched to 5 dB/div.

The amplifier input is about –40 dBm and the output is near –10 dBm indicating a nominal gain of 30 dB (28 dB at the center frequency). The 3 dB cut-off is about 500 MHz, but the response is quite lumpy and rolls off significantly at the low frequency end of the spectrum. In Figure 33 B,C MINUS A mode has been activated. The response is now somewhat smoother because the unflatness of the TG/analyzer system has been subtracted. In this mode the marker can be turned on, and will indicate the gain directly as shown in the figure.

Further tests might include increasing the level of the input signal in 1 dB steps while monitoring the output level for 1 dB increases to determine the 1 dB *compression point* (the point at which the output is 1 dB less than it should be if the output linearly followed the input), or measuring input and output noise to determine the amplifier's noise figure (the amount of noise added by the amplifier).

Return Loss Sweeps. The *return loss* of a device is typically measured to determine how well the device impedance matches a nominal value. The return loss is measured with a spectrum analyzer, *return loss bridge*, and a tracking generator. A return loss bridge designed for the characteristic impedance of the device under test must be used.

The return loss of an antenna is often measured to determine how well it is tuned to the frequency at which it is to transmit or receive. This is possible because antennas achieve their characteristic impedance at their tuned frequency. Figure 34 shows how the equipment is set up to sweep an antenna. In a well constructed and tuned transmission line/antenna system, almost all the power that goes into the system is radiated into space by the antenna and little is reflected. An improperly tuned transmitting antenna can cause much of the energy created by a transmitter to be reflected back into the transmitter resulting in a loss of radiated power, and causing intermodulation distortion. An improperly tuned receiving antenna will result in an apparent loss of receiver sensitivity. The *standing wave ratio (SWR)* is a measure of the transmitted to reflected power ratio of the antenna, and is directly related to the return loss. Figure 35 shows the relationship between return loss, SWR, and reflection coefficient (ρ).

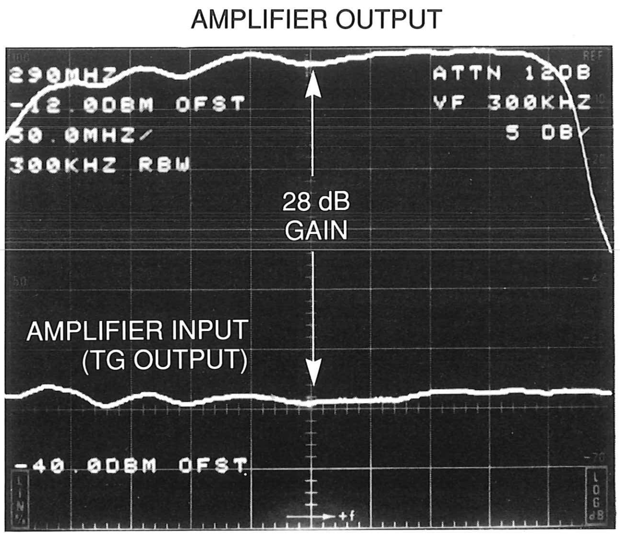


Figure 32. Amplifier frequency response using tracking generator/spectrum analyzer combination.

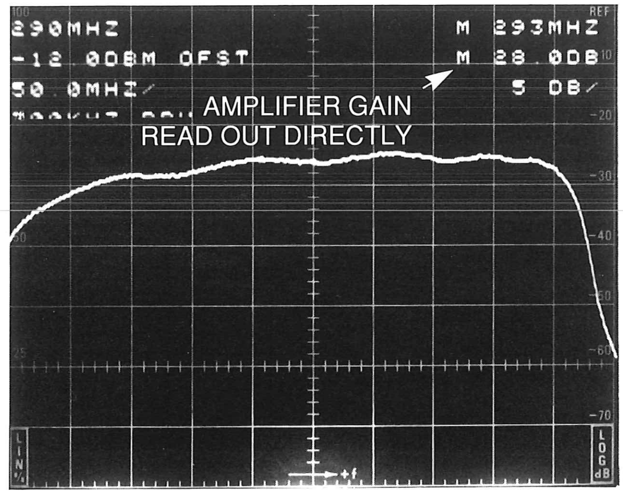


Figure 33. Amplifier frequency response from Figure 32 using B,C MINUS A mode.

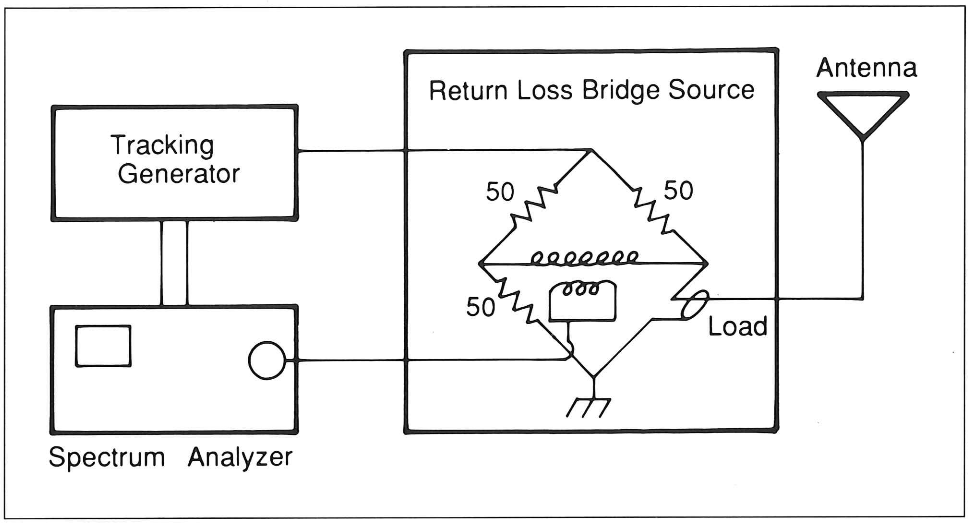
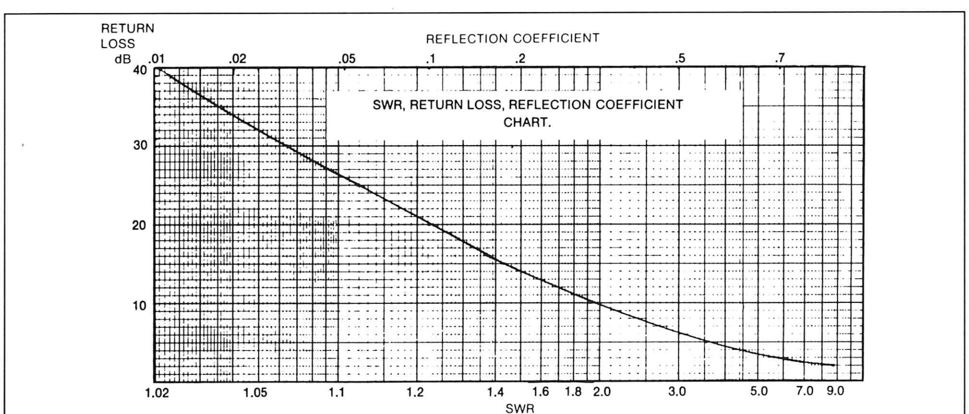


Figure 34. SWR test setup.



$$\mathrm{SWR} = \frac{1 + 10^{-\left(\frac{\text{Return Loss dB}}{20}\right)}}{1 - 10^{-\left(\frac{\text{Return Loss dB}}{20}\right)}}$$

Figure 35. Relationships of return loss, SWR, and reflection coefficient.

When performing antenna sweeps, the analyzer RF attenuator is usually set very low because tracking generator output signals are limited to a few milliwatts. If another nearby transmitter broadcasts while the test is being conducted, excessive power from the transmitter may be coupled into the antenna. This can result in damage to the spectrum analyzer and return loss bridge. If an RF amplifier is available to place between the tracking generator and the return loss bridge, the RF attenuation in the analyzer can be increased. The return loss bridge must then be of a high power variety, and you must check the amplifier for flatness.

During the antenna sweep, the tracking generator sends a signal to the return loss bridge. If the antenna connected to the bridge matches the characteristic impedance that the bridge is designed for, the bridge is balanced; no voltage is developed across its output terminals. If the antenna does not equal the characteristic

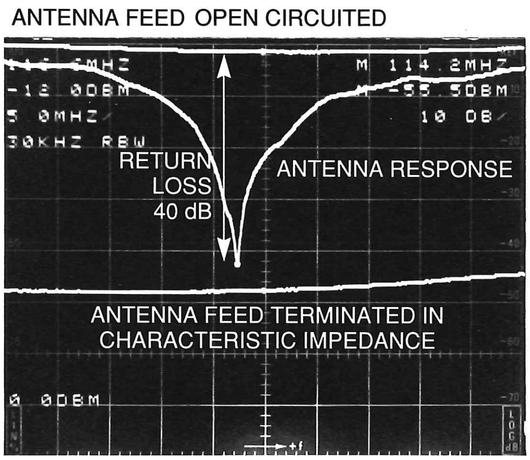


Figure 36. Frequency response of a narrow band antenna.

impedance, the bridge is unbalanced and a voltage is developed across its output terminals which is measured by the spectrum analyzer. As the tracking generator sweeps across the frequency band, the analyzer plots a graph of the bridge output voltage (in dB) vs. frequency.

To perform an antenna sweep, you must first calibrate the system by open-circuiting the antenna end of the transmission line to reflect all the transmitted energy. Adjust the reference level to display the reflected signal near the top of the screen and save the waveform. Next, terminate the antenna end of the transmission line in the characteristic impedance for which the bridge is designed. Repeat the measurement and note the level, which should be quite low (–30 to –60 dB below the open-circuit case). This level should be approached at the antenna resonance. Now connect the antenna to the end of the cable and repeat the measurement again. The difference between the signal level reflected by the antenna and the signal level reflected by the open line is the *return loss.*

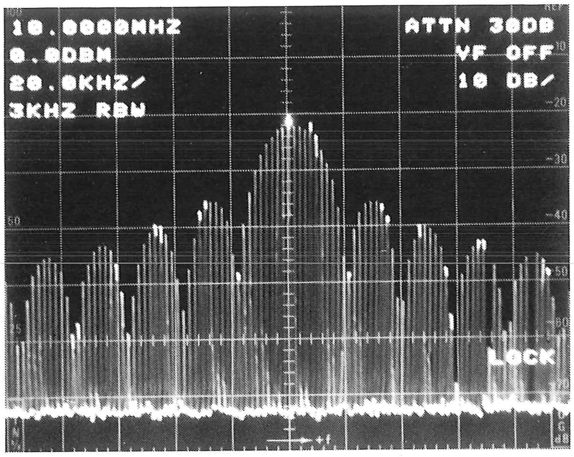
Figure 36 shows an antenna being swept from 92 MHz to 142 MHz. The antenna demonstrates a return loss of about 40 dB at 113 MHz. From the equation in Figure 35 you can determine the antenna's SWR as 1.02:1. At 121 MHz, where the return loss is 10 dB, the SWR is 2.0:1.

For more information on SWR measurements and tracking generator/spectrum analyzer measurements in general, see ***Troubleshooting Two-Way Radios with the Spectrum Analyzer***, Tektronix application note 26AX-3842 and ***The Tracking Generator/Spectrum Analyzer System***, Tektronix application note 26W-5121.

Pulsed Carrier Waveforms

Pulsed carrier waveforms are used in the generation of radar, sonar, CW code, and other signals. These signals turn on a carrier at frequency (f_c) for a fixed period of time (t_{pw}) and then turn it off again. In the case of radar, sonar, and similar signals, the process is usually periodically repeated at a rate of (f_r) pulses per second. The repetition rate is known as the *pulse repetition frequency*, or prf. You can also think of pulsed carrier waveforms as a rectangular signal with pulse width t_{pw} and repetition rate f_r modulating a sine wave carrier with frequency f_c.

How does such a waveform appear when displayed in the frequency domain? With scanning spectrum analyzers, the appearance of the signal's spectrum differs depending on whether the RBW is lesser or greater than the pulse repetition frequency. However, the shape of the envelope of the spectrum does not change. Earlier you learned that all sine waves look the same and that all signals can be described as a combination of sine waves of differing amplitudes. It shouldn't come as too great a shock, therefore, for you to learn that the envelope of the spectra of all rectangular pulsed signals looks the same; it merely gets stretched out or compressed depending on the pulse width – the narrower the pulse, the wider the lobes of the spectrum. In fact, the shape of the envelope is the same as that shown in Figure 4B. Modulating the carrier with the rectangular pulse simply shifts the baseband pulse spectrum to the carrier frequency. Looked at another way, you can view the square wave in Figure 4A as a rectangular signal with pulse width T/2 and repetition rate 1/T modulating a zero frequency carrier.



$$f_r = \frac{1}{t_r}$$

$$t_r = \frac{\text{SWEEP TIME/DIV}}{\text{\# PEAKS/DIV}} = \frac{100\ \text{msec}}{10} = 10\ \text{msec}$$

$$t_{pw} = \frac{1}{\text{LOBE WIDTH}} = \frac{1}{20\ \text{kHz}} = 50\ \mu\text{sec}$$

$$f_c = \text{CENTER FREQUENCY}$$

Figure 37. Spectrum of a pulse waveform.

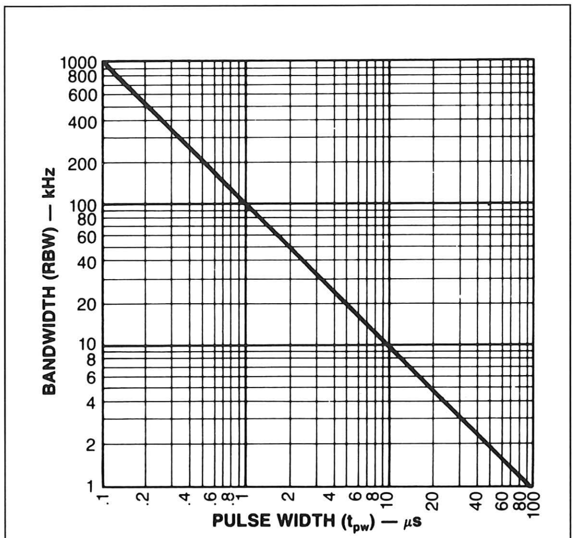


Figure 38. Optimum RBW as a function of pulse width.

In the rest of the discussion of pulsed waveforms, we'll assume the RBW is greater than f_r, which will be true for most applications. The pulse waveform then appears as in Figure 37. The (sin x)/x shape is clearly visible. The annotations in the figure indicate how to determine the pulse width (t_{pw}), repetition rate (f_r), and carrier frequency (f_c) from the spectrum.

But where do all the discrete lines come from? Contrary to what you might think, they do not represent frequency components of the repetitious signal. Those components do exist in theory and are separated by f_r, but can't be resolved because the RBW is much greater than the prf. The lines occur because scanning frequency analyzers look at a particular frequency for only a small portion of the time required to trace the spectrum across the screen. Each time a pulse is generated, the analyzer measures the energy of the pulsed signal at the frequency it is scanning at that instant. If the pulse repetition period (t_r) is 1 ms and the analyzer sweep speed is 1 ms/div, we would see only one line/div. If we slow the sweep speed to 100 ms/div without changing the span/div, we see 100 lines/div because 100 pulses occur while the analyzer sweeps across one division. Remember that although the sweep speed changes, the span of frequencies being analyzed does not (we're just analyzing them more slowly). To compute the repetition rate from Figure 37, count the number of lines per division. Then divide the count by the sweep speed.

To optimize the spectrum of a pulsed waveform, the RBW should be selected narrow enough to display individual lines. However, as the RBW is narrowed, the amount of power from the pulse reaching the detector within the analyzer is reduced and the display will indicate a lower level signal than is actually present. Selecting the correct RBW is a compromise. As a rule of thumb, the RBW should approximately equal 0.1 divided by the pulse width. Figure 38 shows the best RBW as a function of pulse width.

Just as in the case of reducing the RBW, decreasing the pulse width reduces the amount of signal power reaching the detector stage of the analyzer. This also results in an apparent loss of signal strength. The displayed level of a pulsed signal is less than the displayed level of a CW signal of the same voltage. The type of RBW filter used in the analyzer also effects the signal loss. The amount of apparent signal loss between a pulsed waveform and a CW signal of equal peak amplitude is

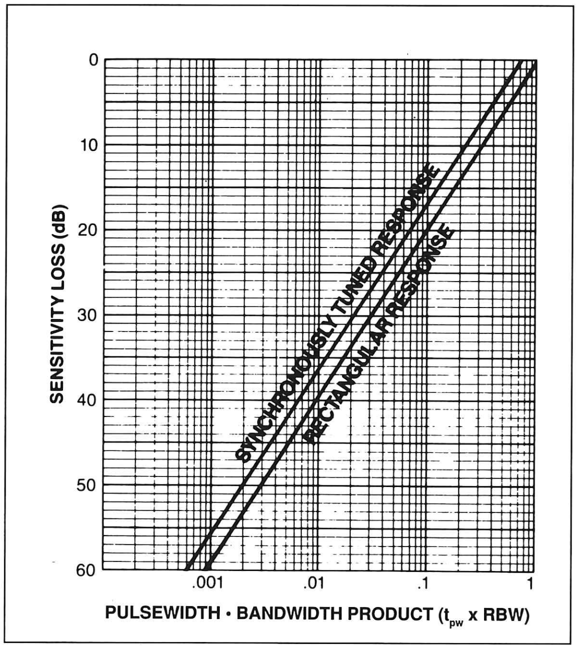


Figure 39. Apparent signal loss of pulsed vs. CW signals as a function of (t_r x RBW).

shown in Figure 39 as a function of *pulse width – RBW product* (t_{pw} x RBW). Tektronix analyzers follow the rectangular response curve even though the actual filter response is not quite rectangular.

Caution

Because there is a signal loss through an analyzer when looking at pulsed signals, it is important to remember that the input of the analyzer may be subjected to larger signals than the screen indicates.

The fast digitizing circuits in newer analyzers like the 2712 enable you to use display storage when viewing the spectra of most pulsed signals. However, it is good practice (especially with lower performance analyzers) to initially disable display storage to ensure no transient signal characteristics are missed. When the desired waveform has been acquired and the optimum combination of sweep speed, span/div, RBW, and reference level have been achieved, the storage facility can be reactivated. Automatic selection of RBW should not be used, because the algorithm used to compute the optimum setting is not valid for pulsed signals. Some spectrum analyzers are equipped with a "pulse stretcher". If yours is so equipped, turn it on when examining pulsed signals.

Here are some general observations you can make from the spectrum of a rectangular pulsed signal:

1. The 1st sidelobe should be approximately 13.3 dB below the main lobe. Other values indicate the pulse is not rectangular.
2. Poorly defined envelope nulls indicate a varying or jittering pulse width.
3. A peak centered atop the main lobe and a void under its center indicate the carrier is not turning off completely (carrier "feed-through").
4. Unsymmetrical lobes indicate the carrier may be FM'ing or jittering.

Caution

Radar applications require relatively large amounts of power for proper operation. Some signal access points on radar systems may have large signal levels that can be lethal to both people and spectrum analyzers. Use Caution and plenty of external attenuation when observing unknown signals.

For further information on pulse measurements, see ***Pulsed RF Spectrum Analysis***, Tektronix application note 26AX-4217-1.

Noise Measurements

A spectrum analyzer is an effective tool for measuring average noise power. Whether you are determining radiated RF noise, the noise figure of an amplifier, or the noise floor of a CATV distribution system, the average noise that you measure is observable as the hashy baseline signal, or "grass", in the spectrum analyzer display. By measuring the difference between the carrier power and the average noise power in a specified bandwidth, you can also determine the *carrier-to-noise (C/N)* ratio.

You must always specify the *noise bandwidth* when measuring noise because noise power is proportional to bandwidth; double the bandwidth and you double the power. If you measure the noise power (in dB) in one bandwidth, you can find the noise power in another from this equation:

$$\text{Noise in new bandwidth} = \text{Noise in old bandwidth} + 10\log\frac{\text{new bandwidth}}{\text{old bandwidth}}$$

This assumes the amplitude of the noise is constant across both bandwidths. Noise power is frequently indicated by units of dBm/Hz or Watts/Hz, which implies that the noise power is what would be measured in a one Hertz bandwidth.

Regardless of the bandwidth selected, it is the **average** noise power which should be measured. If you use a peak display mode without sufficient video filtering, the displayed noise will be far too high. Ensure that you are measuring the average noise.

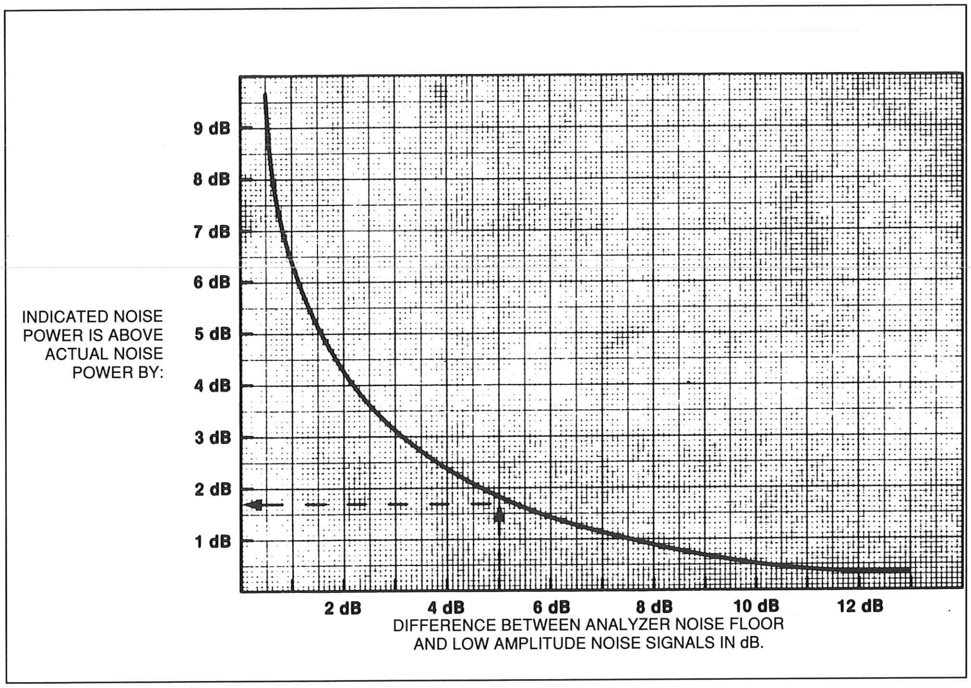


Figure 40. Correction for noise signals within 10 dB of analyzer noise floor.

There are a number of corrections that must be made to the displayed noise level if it is to accurately represent the true noise power. In many analyzers like the 2711 or 2712 the average noise is measured and the corrections are applied automatically – all you do is specify the required noise bandwidth and then place a marker where you want the noise power or C/N measured. Nevertheless, we shall describe the various corrections below so that you understand how noise is measured.

When measuring noise, the noise bandwidth is usually determined by the narrowest filter in the measurement system prior to the detector. In the case of a spectrum analyzer, this is the RBW filter. Because of the filter skirts, the amount of noise passed by the filter is a bit different than what would be predicted from the 3 dB or 6 dB bandwidth. Therefore, a correction factor must be added to the noise measurement to convert from the nominal RBW to the *equivalent noise bandwidth* (bandwidth of a square filter yielding the same measured noise value). Tektronix application note 26W-7045, ***Random Noise Measurement With the Spectrum Analyzer***, explains how the correction can be determined. If the correction is not made, errors of up to 2 dB can occur. Current Tektronix analyzers have the correction factor built in.

The envelope detector and logarithmic amplifier circuits in spectrum analyzers also create differences between the displayed noise level and the true average noise power. The circuits cause the displayed noise to appear lower than the average noise by the following factors:

LIN display mode: 1.05 dB

LOG display mode: 2.5 dB

The algorithms for computing noise power and C/N ratios in Tektronix analyzers compensate for these factors.

Still another potential source of error occurs when measuring noise signals close to the noise floor of the analyzer.

When a noise spectrum is located within 10 dB of the analyzer noise floor, the displayed noise will be higher than the actual value by a factor which you can determine from Figure 40. The reason for this is that at noise levels near the analyzer noise floor, the analyzer self-noise contributes a significant portion of the energy being detected and displayed. Here's another way to look at it: two noise signals that have the same power will indicate 3 dB more than either of the signals separately. Thus, noise that measures 3 dB above the noise floor actually is at the same power as the noise floor. To determine the analyzers noise floor, disconnect all signals from the input and note the level of the noise.

Let's measure the C/N ratio of TV channel 2 using the 2712. You can perform a similar measurement for any channel which has vacant adjacent channel space.

1. Connect the spectrum analyzer to a suitable TV signal source and approximately center the video carrier. Verify the system noise is above the analyzer noise floor by disconnecting from the signal source and noting that the display baseline noise decreases. If it does not, be certain the analyzer's RF attenuator is set to zero and turn on the built-in preamplifier.
2. Activate the Application Menu, specify a 4.0 MHz bandwidth using the Setup Table, and turn on the CARRIER TO NOISE feature.
3. After the spectrum reappears, place the movable marker to the left of the carrier; it indicates where the noise will be measured.

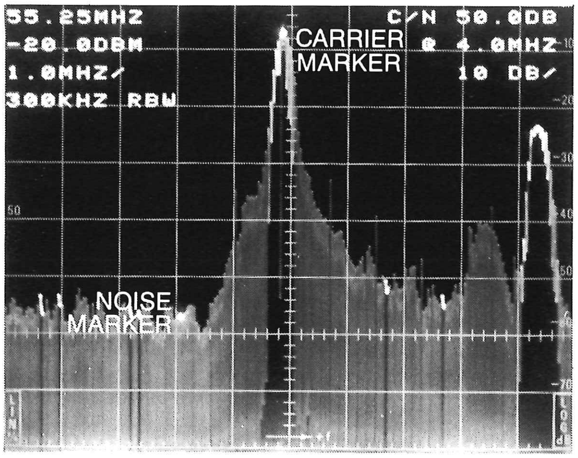


Figure 41. In-service measurement of TV channel 2 C/N ratio.

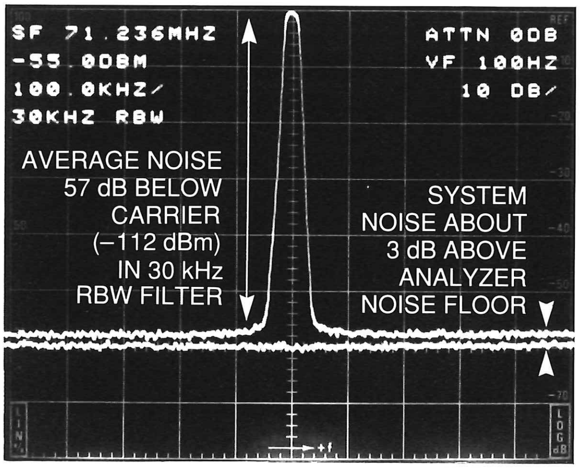


Figure 42. CW carrier-to-noise measurement.

Figure 41 shows a typical display using the 2712's MAX/MIN acquisition mode. The C/N ratio is displayed in the upper right corner of the screen. There is no possibility of arithmetic errors because the analyzer makes all the necessary corrections for you. You can also use any acquisition mode with this analyzer because it measures the average noise power at the marker frequency, but it does not use the displayed noise level.

Suppose now that you must measure a C/N ratio in a 4 MHz bandwidth with a spectrum analyzer that does not have a C/N algorithm. Use the spectra in Figure 42. The average system noise (left-hand or right-hand ends of the upper trace) using a 30 kHz RBW filter is about 3 dB above the noise floor (lower trace) of the analyzer. Find 3 dB on the horizontal scale in Figure 40, and note its corresponding value, also 3 dB, on the vertical scale. A correction factor of 3 dB must be **subtracted** from the indicated noise amplitude to obtain the corrected value. Thus, the actual value is –115 dBm, not the indicated –112 dBm.

This result has to be further corrected for the log error (+2.5 dB) and the RBW-to-equivalent-noise-bandwidth (assume + 1 dB). The final noise value is –111.5 dBm in a 30 kHz bandwidth.

To convert to a 4 MHz equivalent bandwidth, use the equation at the beginning of this section:

Noise in 4 MHz bandwidth =

$$-111.5\text{ dBm} + 10\log\frac{4\text{ MHz}}{30\text{ kHz}} = -90.3\text{ dBm}$$

The carrier peak is at –55 dBm, so the carrier-to-noise ratio is 35.3 dB in the 4 MHz bandwidth.

For more information on the subject of Noise, see ***Noise Measurements Using the Spectrum Analyzer – Part One Random Noise***, Tektronix application note 26AX-3260-1, ***No Loose Ends – Revised: The Tektronix Proof of Performance Program for CATV***, Tektronix application note 26W-7043-1.

Other Application Notes of Interest:

EMI Measurements/Solutions, 26W-7078

EMI Applications Using the 2712, 26W-7071

Glossary

1 dB Compression Point: The point approaching saturation at which the output is 1 dB less than it should be if the output linearly followed input.

Alternating Current (AC): A time-varying electrical signal that switches polarity with the passage of time, often in a periodic fashion (as in powerline voltage).

Amplitude: The magnitude of an electrical signal.

Amplitude Modulation (AM): The process, or result of a process, in which the amplitude of a sine wave (the carrier) is varied in accordance with the instantaneous voltage of a second electrical signal (the modulating signal).

Antenna Sweep: A technique for measuring the return loss of an antenna to determine the antenna tuning.

Audio Frequency Response: The amplitude vs. frequency response of electrical systems or components, such as amplifiers and filters, at audio frequencies.

Average Detection: A detection scheme wherein the average (mean) amplitude of a signal is measured and displayed.

B-SAVE A (or B, C MINUS A): Waveform subtraction mode wherein a waveform in memory is subtracted from a second, active waveform and the result displayed on screen.

Band Switching: Technique for changing the total range of frequencies (band) to which a spectrum analyzer can be tuned.

Base Band: The lowest frequency band in which a signal normally exists (often the frequency range of a receiver's output or a modulator's input); the band from DC to a designated frequency which may be as high as 50 MHz.

Baseline Clipper: A means of blanking the bright baseline portion of the analyzer display.

Bessel Functions: Solutions to a particular type of differential equation. Predicts the amplitudes of FM signal components.

Bessel Null Method: A technique most often used to calibrate FM deviation meters. A modulating frequency is chosen such that some frequency component of the FM signal nulls at a specified peak deviation.

Blocking Capacitor: A capacitor, frequently external to an instrument, which prevents DC signals from passing.

Calibrator: A signal generator producing a specified output used for calibration purposes.

Carrier, Carrier Signal: The electrical signal, typically a sine wave, upon which modulation is impressed.

Carrier Frequency: The frequency of the carrier signal.

Carrier-to-Noise Ratio (C/N): The ratio of carrier signal power to average noise power in a given bandwidth surrounding the carrier; usually expressed in decibels.

Center Frequency: The frequency at the center of a given spectrum analyzer display.

Coax Bands: The range of frequencies that can be satisfactorily passed via coaxial cable.

Comb Generator: A source producing a fundamental frequency component and multiple components at harmonics of the fundamental.

Component: In spectrum analysis, usually denotes one of the constituent sine waves making up electrical signals.

Decibel (dB): Ten times the logarithm of the ratio of one electrical power to another.

Delta F (ΔF): A mode of operation on a spectrum analyzer wherein a difference in frequency may be read out directly.

Diplexers: A device that separates or adds signals at different frequencies.

Direct Current (DC): A constant or very slowly changing electrical signal which does not change polarity.

Distortion: Degradation of a signal, often a result of nonlinear operations, resulting in unwanted signal components. Harmonic and intermodulation distortion are common types.

Dynamic Range: The maximum ratio of two simultaneously present signals which can be measured to a specified accuracy.

Electrical Signal: An electrical current or voltage or some aspect thereof, such as the amplitude or frequency

Emphasis: Deliberate shaping of a signal spectrum or some portion thereof, often used as a means of overcoming system noise. Preemphasis is often used before signal transmission and deemphasis after reception.

Envelope: The limits of an electrical signal or its parameters. For instance, the modulation envelope limits the amplitude of an AM carrier.

Equivalent Noise Bandwidth: The width of a rectangular filter that produces the same noise power at its output as an actual filter when subjected to a spectrally flat input noise signal. Real filters pass different noise power than implied by their nominal bandwidths because their skirts are not infinitely sharp.

External Mixers: A mixer, often in a waveguide format, that is used external to a spectrum analyzer.

Filter: A circuit which separates electrical signals or signal components based on their frequencies.

Filter Loss: The insertion loss of a filter: the minimum difference in dB between the input signal level and the output level.

First Mixer Input Level: Signal amplitude at the input to the first mixer stage of a spectrum analyzer. An optimum value is usually specified by the manufacturer.

Flatness: Unwanted variations in signal amplitude over a specified bandwidth, usually expressed in dB.

Fourier Analysis: A mathematical technique for transforming a signal from the time domain to the frequency domain and vice versa.

Frequency: The rate at which a signal oscillates, or changes polarity, expressed as Hertz or number of cycles per second.

Frequency Band: A range of frequencies that can be covered without switching.

Frequency Band Control: A control used to change frequency bands.

Frequency Control: The device (knob, switch, etc.) on a spectrum analyzer used to change the center or start frequency of the display.

Frequency Deviation: The maximum difference between the instantaneous frequency and the carrier frequency of an FM signal.

Frequency Domain Representation: The portrayal of a signal in the frequency domain; representing a signal by displaying its sine wave components; the signal spectrum.

Frequency Marker: An intensified or otherwise distinguished spot on a spectrum analyzer display indicating a specified frequency point.

Frequency Modulation (FM): The process, or result of a process, in which the frequency of an electrical signal (the carrier) is varied in accordance with some characteristic of a second electrical signal (the modulating signal or modulation).

Frequency Range: That range of frequencies over which the performance of the instrument is specified.

Fundamental Frequency: The basic rate at which a signal repeats itself.

Grass: Noise or a noise-like signal giving the ragged, hashy appear-ance of grass seen close-up at eye level.

Graticule: The calibrated grid overlaying the display screen of spectrum analyzers, oscilloscopes, and other test instruments.

Harmonic Distortion: The distortion that results when a signal interacts with itself, often because of nonlinearities in the equipment, to produce sidebands at multiples, or harmonics, of the frequency components of the original signal.

Harmonic Mixing: A technique wherein harmonics of the local oscillator signal are deliberately mixed with the input signal to achieve a large total input bandwidth. Enables a spectrum analyzer to function at higher frequencies than would otherwise be possible.

Harmonics: Frequency components of a signal occurring at multiples of the signal's fundamental frequency.

Heterodyne Spectrum Analyzer: A type of spectrum analyzer which scans the input signal by sweeping the incoming frequency band past one of a set of fixed RBW filters and measuring the signal level at the output of the filter.

Intermediate Frequency (IF): In a heterodyne process, the sum or difference frequency at the output of a mixer stage which will be used for further signal processing.

IF Gain: The gain of an amplifier stage operating at the IF frequency.

Instantaneous Frequency: The rate of change of the phase of a sinusoidal signal at a particular instant.

Intermodulation Distortion: The distortion that results when two or more signals interact, usually because of non-linearities in the equipment, to produce new signals.

Linear Scale: A scale wherein each increment represents a fixed difference between signal levels.

LO Output: A port on a spectrum analyzer where a signal from the local oscillator is made available. Used for tracking generators and external mixing.

Local Oscillator (LO, L.O.): An oscillator which produces the internal signal that is mixed with an incoming signal in a mixer to produce the IF signal.

Logarithmic Scale: A scale wherein each scale increment represents a fixed ratio between signal levels.

Low Pass Filter: A filter which passes all signal frequencies below a nominal cut off frequency while attenuating all higher frequencies.

Magnitude Only Measurement: A measurement which responds only to the magnitude of a signal and is insensitive to its phase.

MAX HOLD: A spectrum analyzer feature which captures the maximum signal amplitude at all displayed frequencies over a series of sweeps.

Max Span: The maximum frequency span that can be swept and displayed by a spectrum analyzer.

MAX/MIN: A display mode on some spectrum analyzers which shows the maximum and minimum signal levels at alternate frequency points; its advantage is its resemblance to an analog display.

Maximum Input Level: The maximum input signal amplitude which can be safely handled by a particular instrument.

MIN HOLD: A spectrum analyzer feature which captures the minimum signal amplitude at all displayed frequencies over a series of sweeps.

Mixer: A device wherein two or more signals are mixed.

Mixing: The process whereby two or more signals are combined to produce sum and difference frequencies of the signals and their harmonics.

Modulate: To regulate or vary a characteristic of a signal.

Modulating Signal: The signal which modulates a carrier. The signal which varies or regulates some characteristic of another signal.

Modulation: The process of varying some characteristic of a signal with a second signal.

Modulation Envelope: The curve limiting the carrier amplitude of an AM signal.

Modulation Index: In an FM process, the ratio of frequency deviation to the modulating signal frequency.

Noise: Unwanted random disturbances superimposed on a signal which tend to obscure it.

Noise Bandwidth: The frequency range of a noise-like signal. For white noise, the noise power is directly proportional to the bandwidth of the noise.

Noise Floor: The self-noise of an instrument or system that represents the minimum limit at which input signals can be observed. The spectrum analyzer noise floor appears as a "grassy" baseline in the display even when no signal is present.

Noise Sideband: Undesired response caused by noise internal to the spectrum analyzer appearing on the display immediately around a desired response, often having a pedestal-like appearance.

Non-volatile RAM: A type of random access memory (RAM) which does not lose its contents when power is switched off.

Optimum Input Level: Design parameter of the first mixer which allows for maximum dynamic range (largest C/N) and minimum distortion.

Oscillate: To fluctuate systematically with time.

Oscilloscope: An instrument for displaying a time domain representation of an electrical signal; displays the amplitude of an input signal as a function of time.

Peak/Average Cursor: A manually controllable function which enables the user to set the threshold at which the type of signal processing changes prior to display in a digital storage system.

Peak Detection: A detection scheme wherein the peak amplitude of a signal is measured and displayed. In spectrum analysis, 20 log (peak) is often displayed.

Peaking: The process of adjusting a circuit for maximum amplitude of a signal by aligning internal filters. In spectrum analysis, peaking is used to align preselectors.

Percent Amplitude Modulation: A relative measure of how much a carrier is amplitude modulated; 100 times the ratio of the modulation envelope amplitude to the amplitude of the unmodulated carrier.

Period: The time interval at which a process recurs; the inverse of the fundamental frequency.

Phase Lock: The control of an oscillator or signal generator so as to operate at a constant phase angle relative to a stable reference signal source. Used to ensure frequency stability in spectrum analyzers.

Preselector: A tracking filter located ahead of the first mixer which allows only a narrow band of frequencies to pass into the mixer.

Primary Controls: Those controls of fundamental importance to spectrum analyzer operation: frequency control, span/div, reference level.

Products: Signal components resulting from mixing or from passing signals through other non-linear operations such as modulation.

Pulse Stretcher: Pulse shaper that produces an output pulse whose duration is greater than that of the input pulse and whose amplitude is proportional to that of the peak amplitude of the input pulse.

Pulse Repetition Frequency (PRF): The frequency at which a pulsing signal recurs: equals the fundamental frequency of the pulse train.

Pulsed Carrier Waveforms: Signal types wherein the carrier is turned on and off, or pulsed, as in radar, sonar, and CW code.

Reference Level: The signal level required to deflect the CRT display to the top graticule line.

Reference Level Control: The control used to vary the reference level on a spectrum analyzer.

Resolution Bandwidth (RBW): The width of the narrowest filter in the IF stages of a spectrum analyzer. The RBW determines how well the analyzer can resolve or separate two or more closely spaced signal components.

Resolution Bandwidth (RBW) Control: On a spectrum analyzer the control used to vary the resolution bandwidth, usually by selecting alternate RBW filters.

Return Loss: The ratio of power sent to a system to that returned by the system. In the case of antennas, the return loss can be used to find the SWR.

Return Loss Bridge: A device used in measuring return loss.

RF Attenuator: An attenuator built and calibrated for use up to and including radio frequencies.

RF Power Meter: An instrument used to measure signal power at radio frequencies.

Ring: A transient response wherein a signal initially performs a damped oscillation about its steady-state value.

SAVE Function: A feature of spectrum analyzers incorporating display storage which enables them to store displayed spectra.

Scanning Spectrum Analyzer: See heterodyne spectrum analyzer.

Sensitivity: Measure of a spectrum analyzer's ability to display minimum level signals at a given IF bandwidth, display mode, and any other influencing factors.

Shape Factor: In spectrum analysis, the ratio of an RBW filter's 60 dB bandwidth to its 3 dB or 6 dB width (depending on manufacturer).

Sidebands: Signal components observable on either or both sides of a carrier as a result of modulation or distortion processes.

Sideband Suppression: An amplitude modulation technique wherein one of the AM sidebands is deliberately suppressed, usually to conserve bandwidth.

Signal Identifier: A method and control for distinguishing the desired signal from unwanted mixer outputs. Typically used with a mixing process for which preselection cannot be implemented.

Sine Wave: A smoothly-varying oscillatory signal with a characteristic sinusoidal shape. All periodic signals can be represented as a sum of sine waves of differing frequencies.

Single Sweep: Operating mode in which the sweep generator must be reset for each sweep. Especially useful for obtaining single examples of a signal spectrum.

Span/Div Control: The control used to vary the span/div of a spectrum analyzer.

Span Per Division, Span/Div: Frequency difference represented by each major horizontal division of the graticule.

Spectrum: The frequency domain representation of a signal wherein it is represented by displaying its frequency distribution.

Spectrum Analysis: The technique or process of determining the frequency distribution of a signal.

Spectrum Analyzer: A device for determining the frequency components of a signal.

Spurious Response: An undesired extraneous signal produced by mixing, amplification, or other signal processing technique.

Square Wave: A pulsing rectangular signal with equal on and off times.

Stability: The property of retaining defined electrical characteristics for a prescribed time and in specified environments.

Starting Frequency: The frequency at the left hand edge of the spectrum analyzer display.

Suppressed Carrier: An amplitude modulation technique wherein the carrier is deliberately suppressed. Often combined with side-band suppression to produce single sideband suppressed carrier signals for communications purposes.

Sweep Control: The control or controls used to vary the sweep parameters. The most important control in this group is sweep speed.

Sweep Speed, Sweep Rate: The speed or rate, expressed in time per horizontal divisions, at which the electron beam of the CRT sweeps the screen.

Standing Wave Ratio (SWR): The ratio of the maximum amplitude to the minimum amplitude of a signal along a transmission line caused by signal reflections. An impedance mismatch causes reflections of an outgoing signal to combine with the signal itself to produce a series of "standing," or stationary, peaks and nulls.

Total Harmonic Distortion (THD): Distortion due to all harmonics as opposed to just the first, or second, etc. Ratio of the rms sum of all distortion products to the fundamental.

Time Domain Representation: Representation of signals by displaying the signal amplitude as a function of time. Typical of oscilloscope and waveform monitor displays.

Time-Varying Signals: A signal whose amplitude changes with time.

Total Span: The total width of the displayed spectrum. The span/div times the number of divisions.

Tracking Generator: A signal generator whose output frequency is synchronized to the frequency being analyzed by the spectrum analyzer.

Ultimate Rejection: The ratio, in dB, between a filter's passband response and its response beyond the slopes of its skirts.

Video Filter: A postdetection low pass filter (sometimes referred to as a noise averaging filter) used primarily to remove noise from the displayed spectrum.

Vertical Scale Factor, Vertical Display Factor: The number of dB, volts, etc. represented by one vertical division of a spectrum analyzer display screen.

Video Filter Control: The control used to vary the video filter bandwidth, usually by selecting an alternate filter, and to turn it on or off.

VIEW Function: A feature and/or control which enables or disables a waveform memory for viewing.

Waveform Memory: Memory dedicated to storing a digital replica of a spectrum.

Waveform Subtraction: A process wherein a saved waveform can be subtracted from a second, active waveform.

Waveguide Bands: The range of frequencies in which waveguide technology (as opposed to coaxial cable technology) is used for signal transmission.

Zero Hertz Peak: A fictitious signal peak occurring at zero Hertz that conveniently marks zero frequency. The peak is present regardless of whether or not there is an input signal.

Zero Span: A spectrum analyzer mode of operation in which the RBW filter is stationary at the center frequency; the display is essentially a time domain representation of the signal propagated through the RBW filter.

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