Source Basics

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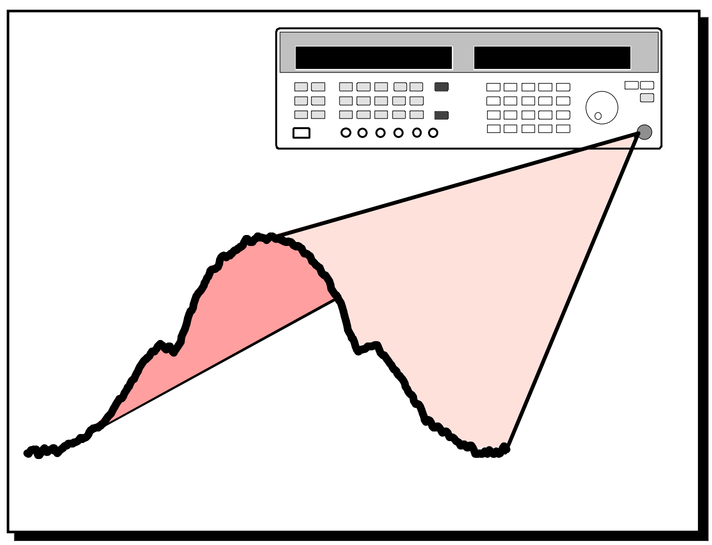

1997 Back to Basics Seminar

Abstract

To prepare you for the challenges of today's signal generation, we'll cover the basics on the signals required to test a variety of products from amplifiers to highly secure communication systems. These signals may be as simple as a single frequency sinusoid or as complex as a digitally modulated carrier. This seminar reviews the basics of signal generators and the applications where signal generators are used. Block diagrams will be reviewed where appropriate. Signal generator specifications will also be discussed.

Author

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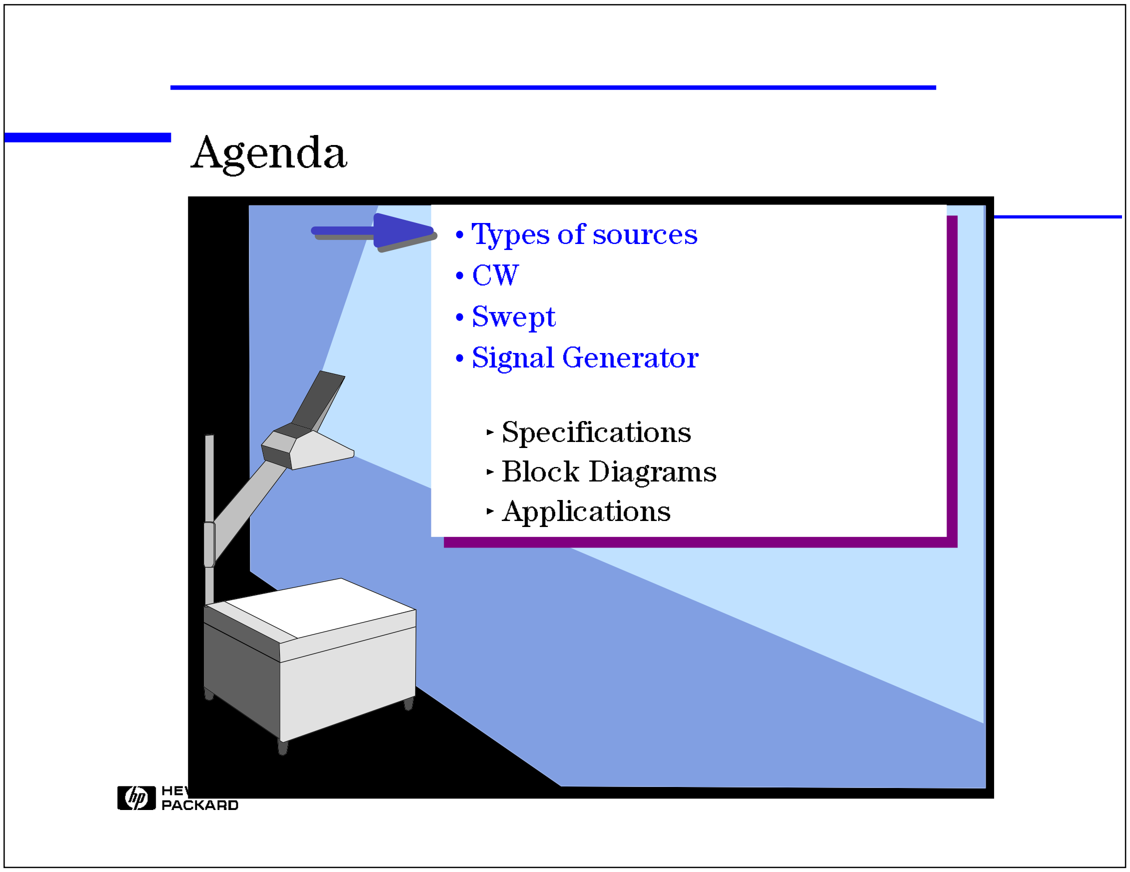
Slide #1



A signal source produces sine waves. This is the most basic definition of a signal source.

In this seminar, three types of sources will be reviewed. Block diagrams will be used to explain how sources work. For each type of source, several applications will be reviewed and the critical specifications for each application will be emphasized.

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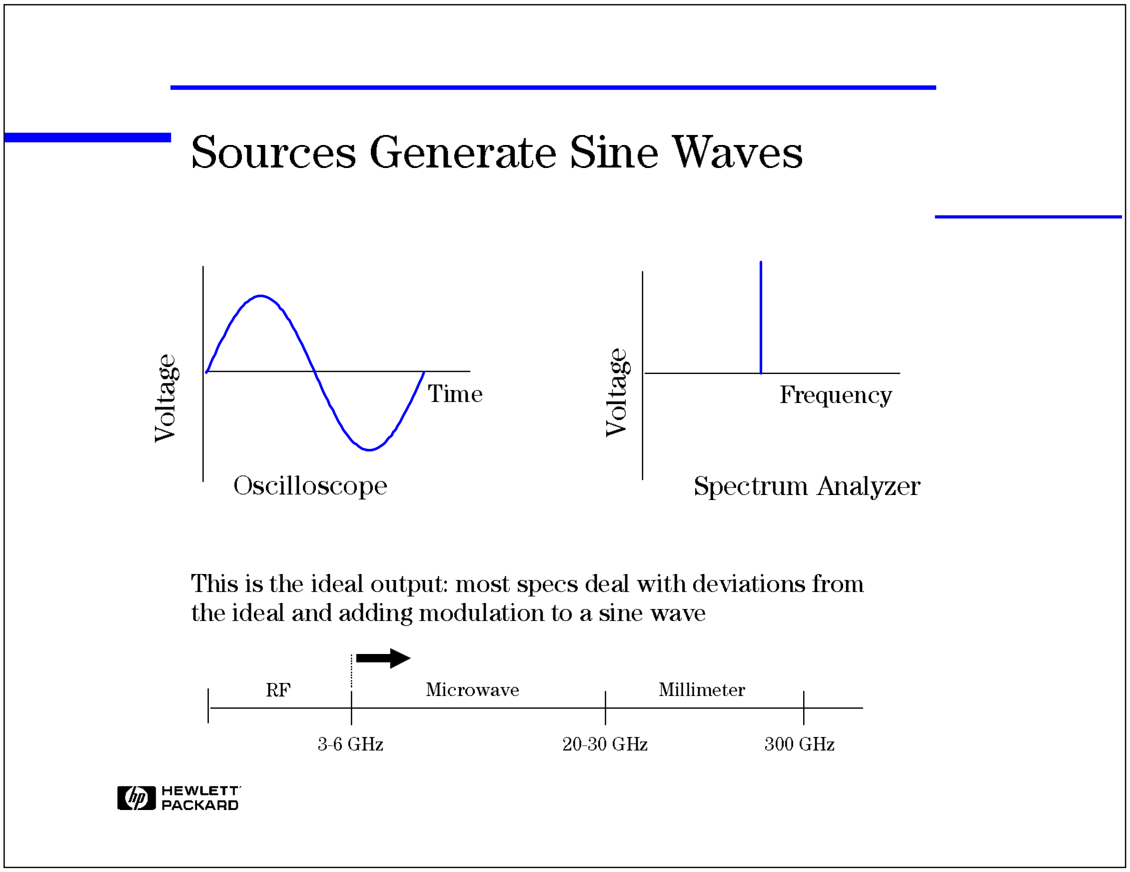

Sources generate sine waves. Sine waves are used in many, many test and measurement applications. The ideal output is shown in both the time domain and the frequency domain. The time domain waveform is expressed by:

$$v(t) = v_o \sin(2\pi f_o t)$$

Using Fourier analysis, the time domain waveform may be transformed to a frequency domain representation:

$$\begin{aligned} V(f) &= \int_{-\infty}^{+\infty} v(t) e^{-j2\pi ft} dt \\ &= \int_{-\infty}^{+\infty} v_o \sin(2\pi f_o t) e^{-j2\pi ft} dt \\ &= -jv_o/2[\delta(f - f_o) - \delta(f + f_o)] \end{aligned}$$

Where the function $\delta(\,)$ is the impulse function. The Fourier transform is a two-sided representation with impulse functions centered at both f_o and $-f_o$. The amplitude of each impulse function is $v_o/2$. The equivalent single sided representation is a single impulse function at f_o with an amplitude of v_o.

The performance specifications for sources identify the range of output power and frequencies available. In addition, specifications identify deviations from the ideal sine wave. These deviations can be intentional, such as adding modulation capabilities, or unintentional.

Slide #3

Three basic types of sources will be discussed: CW sources, swept sources, and signal generators. A source that produces a single sine wave is referred to as a CW source. The frequency and the amplitude of the sine wave can be set to a desired value in most CW sources. A swept source adds the ability to automatically vary the output frequency or amplitude of a sine wave over a range of frequencies or amplitudes. Some swept sources have the ability to vary frequency and amplitude simultaneously. When modulation is added to a sine wave, the source is a signal generator. Signal generators output "signals" : Sine waves that carry information. There are numerous methods for adding information to a sine wave. Basics signal generators have amplitude, frequency, and phase modulation capabilities. More advanced signal generators have pulse and IQ modulation capabilities.

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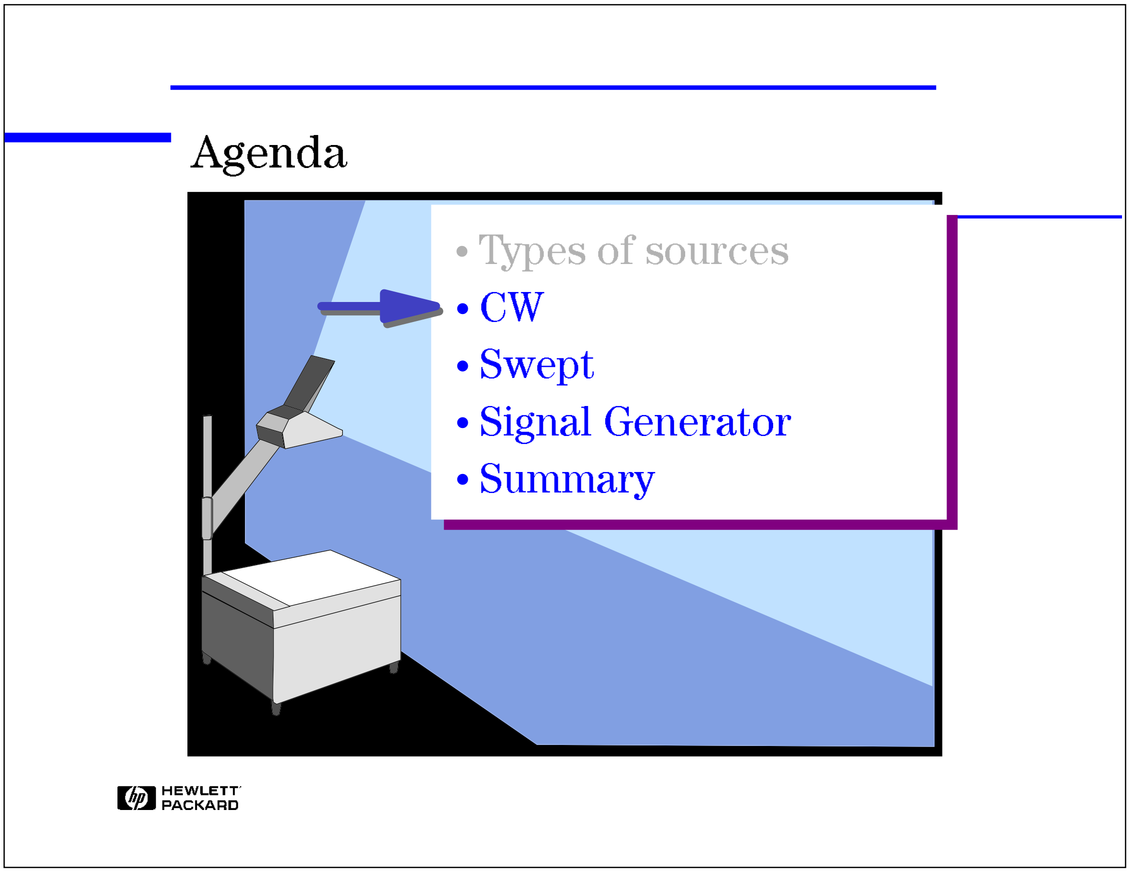
Agenda
• Types of sources
• CW
• Swept
• Signal Generator
• Summary
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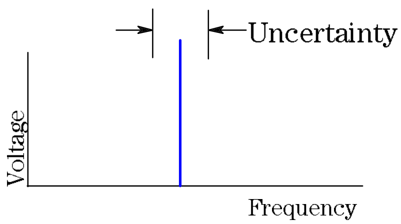
Slide #5

CW Source Specifications

...Frequency

- Range: Range of frequencies covered by the source
- Resolution: Smallest frequency increment
- Accuracy: How accurately can the source frequency be set
- Switching Speed: Automated test applications



EXAMPLE

Accuracy = $\pm f_{CW} * \tau_{aging} * \tau_{cal}$

f_{CW} = CW frequency = 1 GHz
τ_{aging} = aging rate = 0.152ppm/year
τ_{cal} = time since last calibrated = 1 year

Accuracy = $\pm$ 152 Hz

Understanding source specifications is critical when determining the appropriate source for an application. For CW sources, the specifications are generally divided into three broad categories: Frequency, amplitude (or output), and spectral purity.

Range, resolution, and accuracy are the main frequency specifications. Range specifies the range of output frequencies that the source can produce. Resolution is the smallest frequency increment. The accuracy of a source is affected by two parameters: The stability of the reference oscillator and the amount of time that has passed since the source was last calibrated. A typical (but very good) reference oscillator may have an aging rate of 0.152ppm (parts-per-million) per year. The aging rate indicates how far the reference will drift (either up or down) from its specified value. At 1 GHz, a source that has not been calibrated for one year with an aging rate of 0.152ppm per year will be within 152 Hz of its specified output frequency.

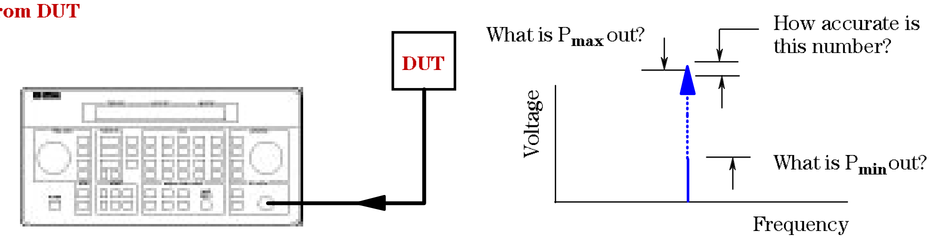
Slide #6

CW Source Specifications

...Amplitude

- Range (-136dBm to +13dBm)
- Accuracy (+/- 0.5dB)
- Resolution (0.02dB)
- Switching Speed (25ms)
- Reverse Power Protection

Source protected from accidental transmission from DUT



Range, accuracy, resolution, switching speed, and reverse power protection are the main amplitude specifications. The range of a source is determined by the maximum output power and the amount of internal attenuation built into the source. Sources monitor their own output power to maintain amplitude accuracy. Automatic leveling circuits are used to measure the output. The resolution of a source indicates the smallest amplitude increment. Switching speed is a measure of how fast the source can change from one amplitude level to another.

Sources are often used to test transceivers. Because transceivers have transmitters, the connection between a source and the transceiver could conduct a signal from the device being tested to the output connector of the source. Reverse power protection prevents signals traveling the wrong direction from damaging the source.

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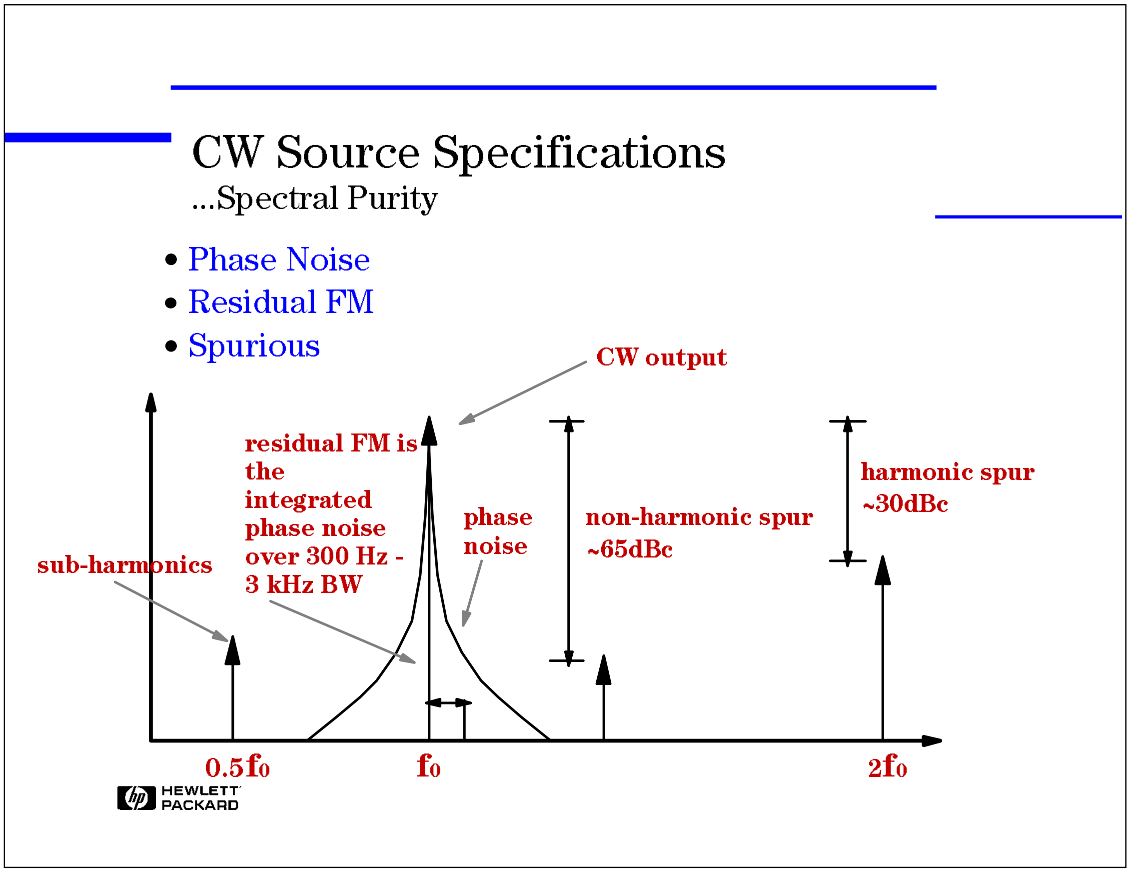

The specifications associated with spectral purity are often the most difficult to understand. The ideal CW output is a sine wave at a single frequency. Unfortunately, their are no ideal CW sources: All sources are made with non-ideal (i.e. real) components. These components introduce phase noise and unwanted distortion products.

Harmonic spurs are integer multiples of the CW output. Sources contain many non-linear components. These components are needed to provide a broad range of frequencies and output powers. Consider the output of an amplifier:

$$v_o(t) \quad = \quad a_1\, v_i(t) + a_2\, v_i^2(t) + a_3\, v_i^3(t) + \ldots$$

For an input sine wave, the output is:

$$v_o(t) \quad = \quad a_1 \sin(\omega t) + a_2 \sin^2(\omega t) + a_3 \sin^3(\omega t) + \ldots$$

$$= \quad a_2/2 + a_1 \sin(\omega t) + 3a_3/4 \sin(\omega t) + \; a_2/2 \sin(2\omega t) + a_3/4 \sin(3\omega t) + \ldots$$

The non-linear characteristics of an amplifier create second, third, and higher order harmonics. A typical second harmonic will be specified at <-30 dBc (better than 30 dB below the output of the fundamental frequency). Non-harmonic spurs come from a variety of sources (e.g. power supply) and are typically quite low (<-65 dBc).

Multipliers are often used in sources to extend the frequency output. This results in the presence of sub-harmonics.

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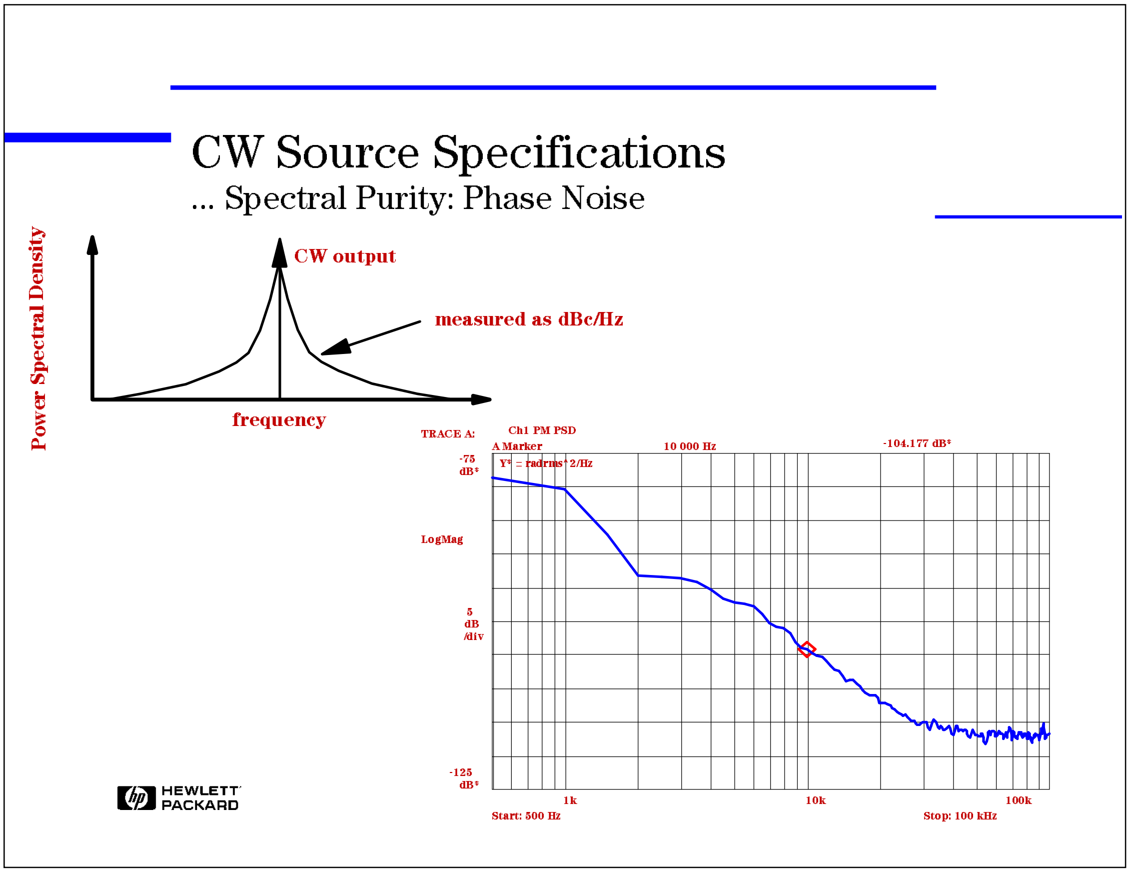

In the ideal case, all of the power in a sine wave is concentrated at a single frequency. Random noise within the source will cause the power to be spread over a small range of frequencies. The spread is referred to as phase noise and is often mathematically modeled as random phase modulation. The units of phase noise are dBc/Hz: dB down from the carrier in a 1 Hz bandwidth. Phase noise is specified at a frequency offset from the CW output. For example, the phase noise of a CW source may be specified as: -97dBc/Hz @ 100 kHz offset from a CW frequency at 20 GHz.

Phase noise may be directly measured from the spectrum of a source. This method requires that the phase noise of the analyzer be much better (~10dB) than the phase noise of the source being tested. Often, the phase noise of a source is measured using test equipment that has been optimized for this purpose. Phase noise is generally displayed on a log-log axis. This enables both the close in phase noise (offsets < 1 kHz) and the far out phase noise (offsets > 10 kHz) to be easily examined on one plot.

The phase noise plot above was generated using the HP 89441A by displaying the power spectral density (units of dBm/Hz) of a phase demodulated signal. The source is at 1 GHz. The marker is at a 10 kHz offset and reads -104dBc/Hz.

Residual FM is a measure of the small amount of FM inherent in an CW output. Residual FM is specificied within a bandwidth. Most sources typically specify residual FM per the CCITT specified bandwidth. The CCITT bandwidth starts at 300 Hz offset from the carrier frequency and stops at a 3 kHz offset. Within this band, all of the noise shown on the phase noise curve contributes to residual FM. ..

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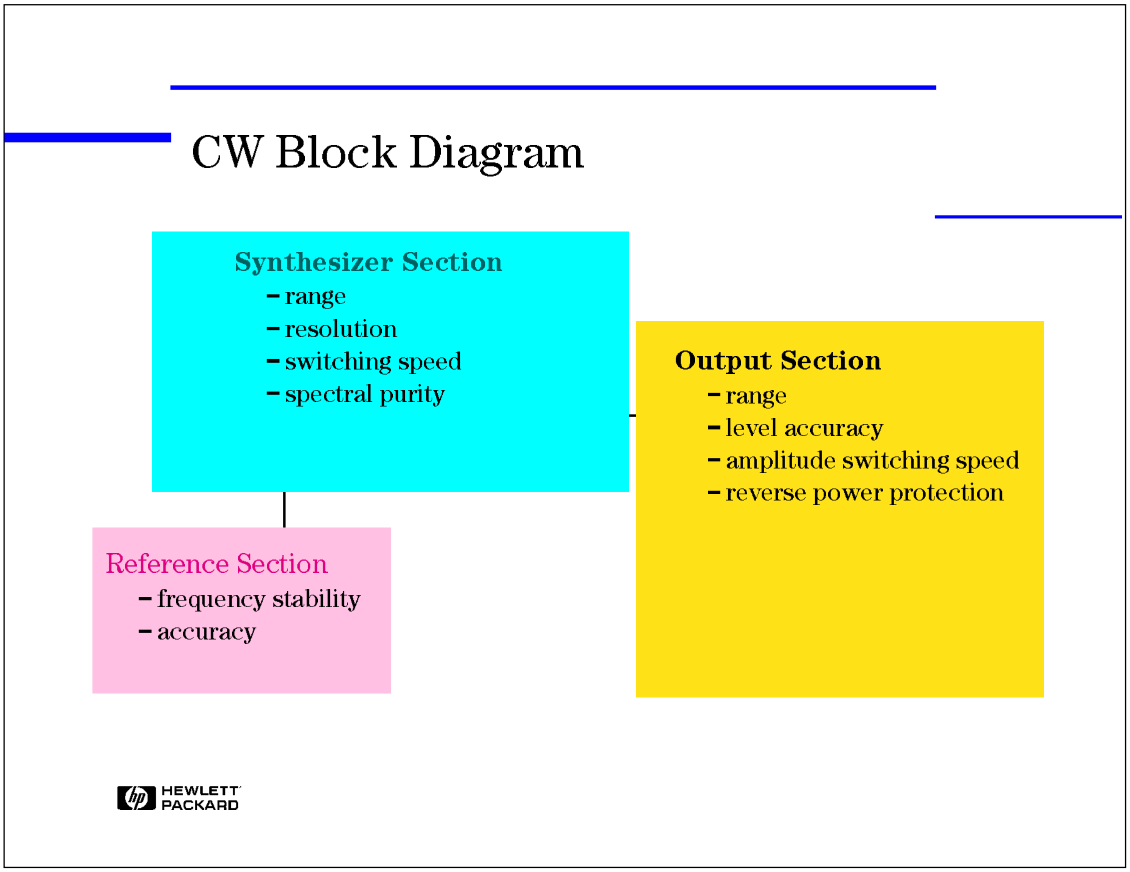
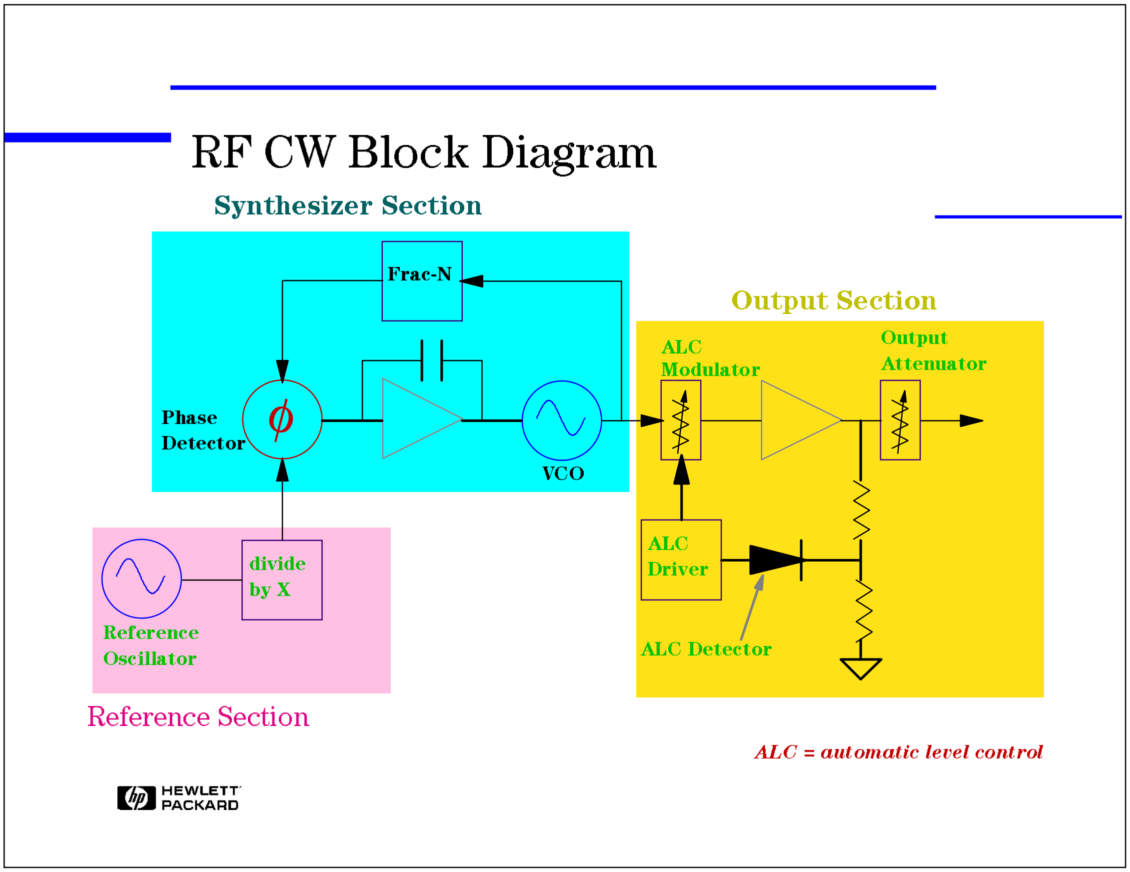


The block diagram of a CW source can be divided into three major sections: Reference, synthesizer, and output. Each section has a unique role in producing a sine wave and makes a unique contribution to the source specifications.

Slide #10



The above block diagram provides greater detail for an RF CW source.

The reference section supplies the reference oscillator for the source. The reference oscillator contributes to the short term stability of the output frequency (phase noise). The long term stability of the reference oscillator, the aging rate, determines the accuracy of the output frequency.

The reference section supplies a sine wave with a known frequency to the synthesizer section. This sine wave is used as the reference for a phase-locked loop (PLL). The synthesizer section is responsible for producing a clean sine wave at the desired frequency. The VCO (voltage controlled oscillator) produces the sine wave. The PLL maintains the output frequency at the desired setting and translates the frequency accuracy of the reference oscillator to the output of the VCO.

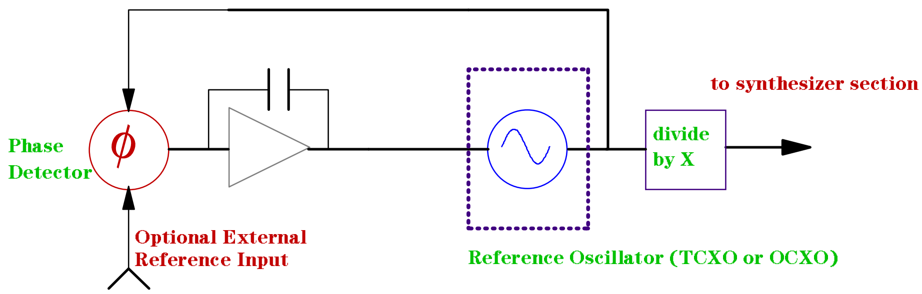
The synthesizer section supplies a clean sine wave to the output section. The output section determines the overall amplitude range and accuracy of the source. Amplitude range is determined by the available amplification and attenuation. Amplitude accuracy, or level accuracy, is maintained by monitoring the output power and adjusting the power as needed.

Let's take a look at each section in more detail.

Slide #11

RF CW Block Diagram

Reference Section



	TCXO	OCXO
Aging Rate	+/- 2ppm/year	+/- 0.1 ppm /year
Temperature	+/- 1ppm/year	+/- 0.01 ppm/year
Line Voltage	+/- 0.5ppm/year	+/- 0.001 ppm/year

The heart of the reference section is the reference oscillator. The reference oscillator must be inexpensive, extremely stable and adjustable over a narrow range of frequencies. A stable reference oscillator will ensure that the frequency output of the source remains accurate in between calibrations. By comparing the reference oscillator to a frequency standard, such as a Cesium oscillator, and adjusting as needed, the source can be calibrated with an output that is traceable.

Of all materials today, crystalline quartz best meets these criteria. The fundamental frequency of quartz is affected by several parameters: aging, temperature, and line voltage. Over time, the stress placed on a quartz crystal will affect the oscillation frequency. Temperature changes cause changes in the crystal structure which affect the oscillation frequency. The piezoelectric nature of quartz is also affected by the electric fields created inside the source by the line voltage.

To improve the performance of quartz, temperature compensation circuitry is used to limit the variations in output frequency that result from variations in the operating temperature. Crystals with such compensation are referred to as **T**emeprature **C**ompensated crystal oscillators or TCXO's. OCXO's are crystals that have been placed in an **O**ven **C**ontrolled environment. This environment maintains a constant temperature and provides shielding from the affects of line voltage. The stability for both TCXO's and OCXO's is tabulated above.

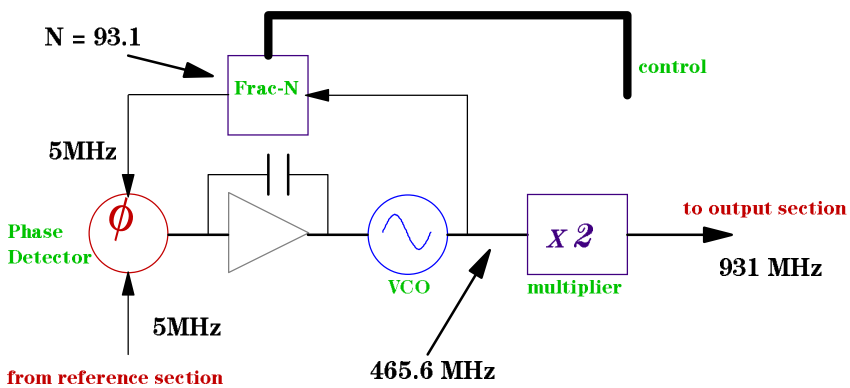
Many sources provide an external input that may be used to lock the oscillator to an external reference. The source, however, does not require an external reference.

Slide #12

RF CW Block Diagram

Synthesizer Section

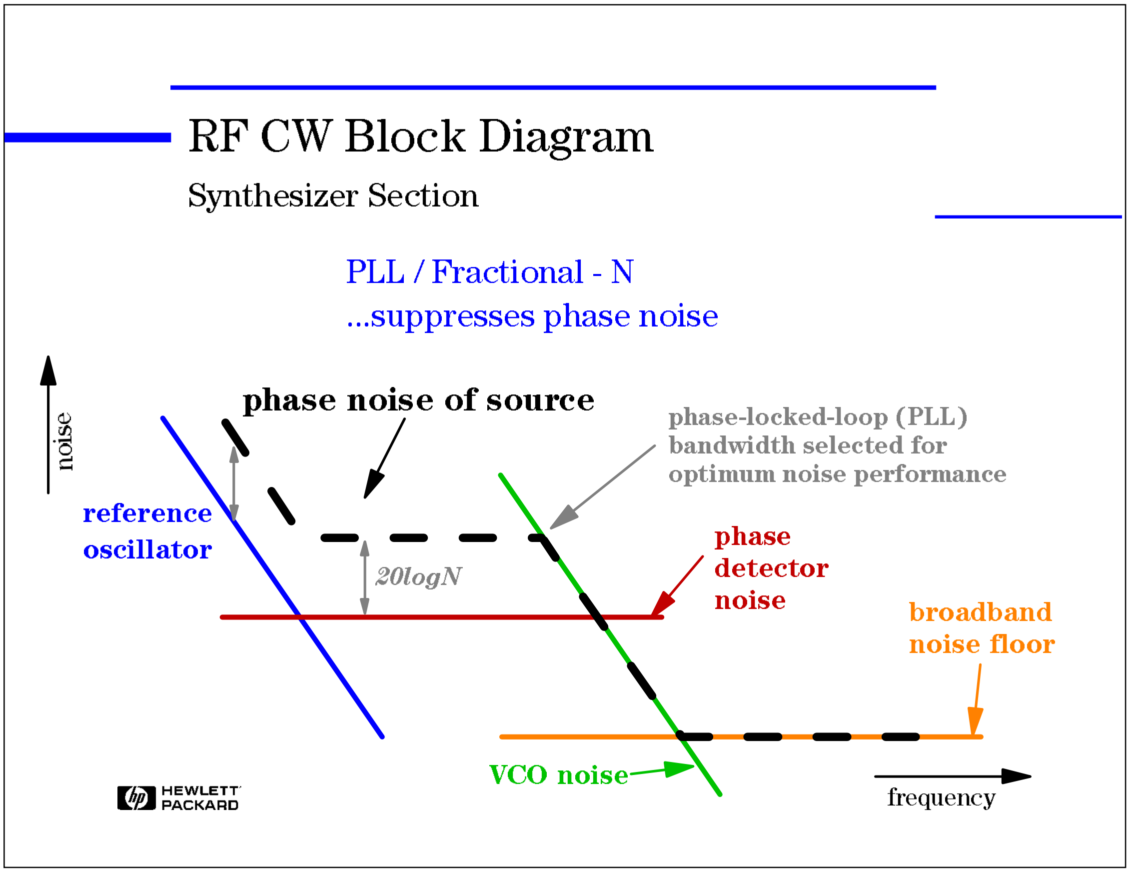
...produces accurate, clean signals



A VCO produces an output frequency for an input voltage. A simple VCO can be constructed from a varactor, a voltage-variable capacitor. A reverse-biased pn junction diode is a common type of varactor. The capacitance across a diode decreases as the reverse bias to the diode increases. When placed in an oscillator circuit, the tunable capacitor enables the output oscillation to be tuned. This output is inherently unstable. A PLL is required to maintain frequency stability.

Most of the VCO output is sent to the output section of the source. A portion of the VCO output is divided to a lower frequency. In the example above, the VCO output at 465.5 MHz is divided by 93.1 to produce a frequency of 5 MHz. This signal is compared to the 5 MHz signal supplied by the reference section. The output of the phase detector will be a dc offset with an error signal. The dc offset represents the constant phase difference between the 5 MHz signal from the reference and the 5 MHz signal from the Frac-N divide circuit. The error signal represents unwanted frequency drift. The output of the phase detector is filtered and amplified to properly drive the VCO. If the VCO does not drift, there will be (almost) no error signal at the output of the phase detector and the control voltage to the VCO will not change. If the VCO drifts upwards (or downwards), the error signal at the output of the phase detector will adjust the VCO output downwards (or upwards) to maintain a stable frequency output.

Slide #13



The synthesizer section of a source has a tremendous impact on the overall phase noise of the source. There are four main contributors to phase noise: The reference oscillator, the phase detector, the VCO, and the broadband noise floor. The broadband noise floor results primarily from the thermal noise present in the source. In general, this noise does not greatly limit the performance of the source. The phase noise of the reference oscillator and the VCO both fall off initially as $1/f^3$ and transitions to a $1/f^2$ dependence. On a log-log plot, $1/f^2$ translates to a slope of 20 dB per decade. The phase noise contribution of the phase detector is dominated by thermal noise and hence exhibits the same spectral dependency (or lack of spectral dependency) as the broadband noise floor. In addition, the Frac-N divide in the PLL degrades the phase noise performance by 20logN where N is the divide by number.

The bandwidth of the PLL determines the point at which the VCO contribution to the overall phase noise becomes supressed. For frequency offsets inside the PLL bandwidth, the overal phase noise of the source is dependent mainly upon contributions from the phase detector and the reference oscillator.

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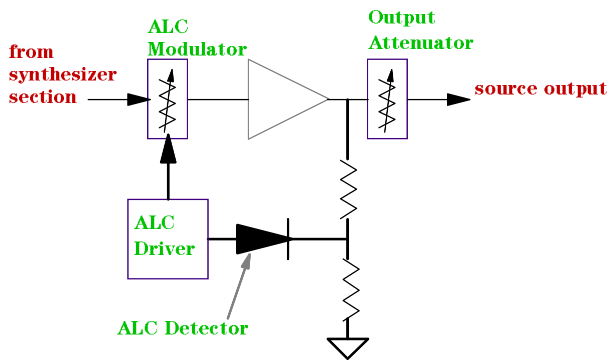
Slide #14

RF CW Block Diagram

Output Section

- ALC
  - maintains output power by adding/subtracting power as needed
- Output Attenuator
  - mechanical or electronic
  - provides attentuation to achieve wide output range (e.g. -136dBm to +13dBm)



ALC = automatic level control

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The output section maintains amplitude or level accuracy by measuring the output power and compensating for deviations from the set power level. The ALC driver digitizes the detector output and compares the digitized signal to a look-up table. The appropriate modulator drive is generated such that the detected power becomes equivalent to the desired power. Frequently, external losses from cabling and switching between the output of the signal source and the device under test (DUT) attenuate the signal. A look-up table that compensates for external losses can be input to extend the automatic leveling to the input of the DUT.

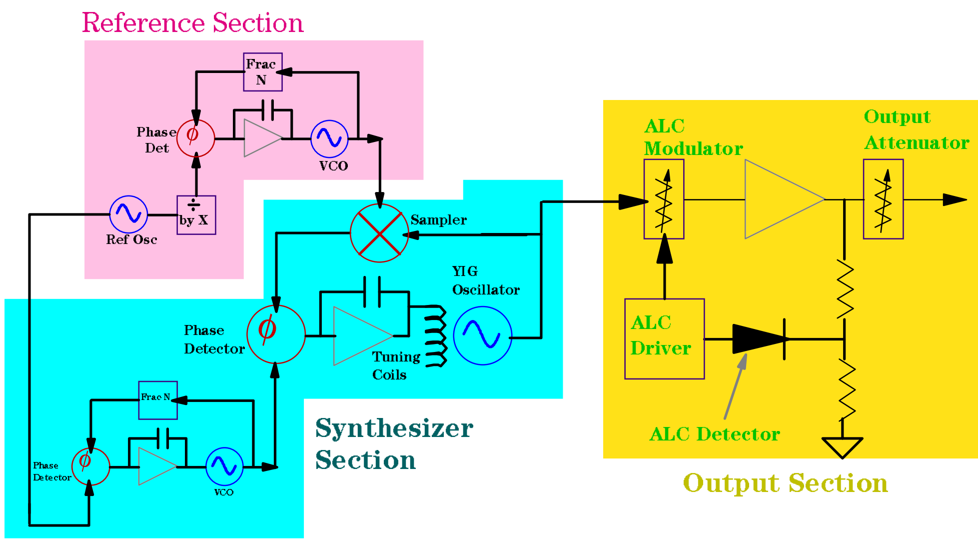
With no output attenuation applied, the source amplitude is at a maximum. The maximum amplitude is determined by the power amp and the loss between the output of the amp and the output connector. The main source of loss is the output attenuator. The output attenuator will introduce a finite amount of loss even when the attenuation is set to 0 dB. The purpose of the output attenuator is to reduce the output power in a calibrated and repeatable fashion. Today attenuators are available that provide output ranges from +13 dBm (no attenuation applied to the source) to -136 dBm (maximum attenuation applied). There are two types of attenuators that are commonly used: mechanical and solid state.

Mechanical attenuators introduce very little loss between the output of the power amp and the output connector. Thus a high output power can be achieved without over-driving the output amplifier. Operating at low drive levels reduces the level of harmonics generated by the source. Mechanical attenuators do, however, have finite lifetimes. A typical mechanical attenuator will live for five million cycles. For an ATE application in which the power level is changed every two seconds, the attenuator will fail after about a year.

Solid state or electronic attenuators have essentially infinite lifetimes. For ATE applications, solid state attenuators are well suited. Solid state attenuators do introduce significant loss even when no attenuation is desired. Higher output amplifier drive levels are required in sources with solid state attenuators to overcome the losses. The higher drive levels increase the level of the harmonics. Sources with solid state attenuators, therefore, require more sophisticated designs to maintain an equivalent level of spectral purity with sources that use mechanical attenuators.

Slide #15

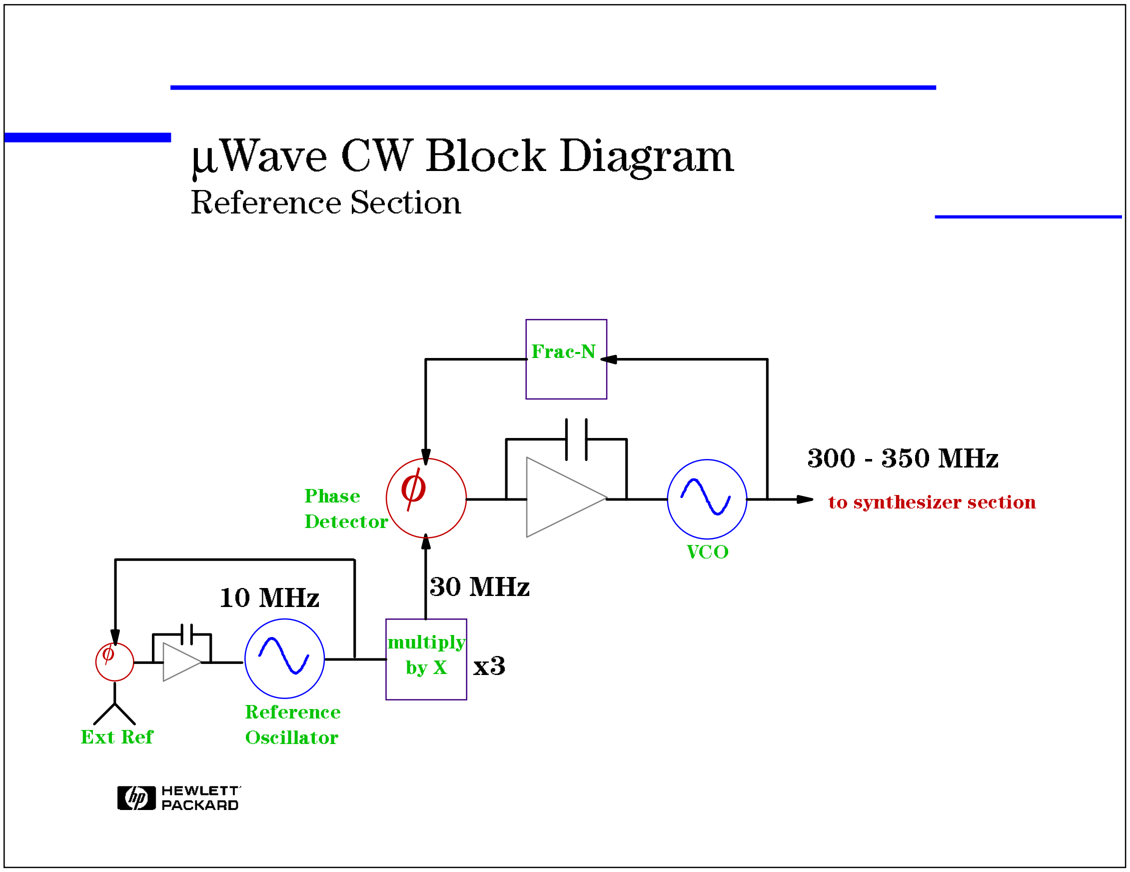
µWave CW Block Diagram



The block diagram of a microwave CW source is similar to that of an RF CW source; each has the same three basic sections. There are differences, however. Although the reference section only has one reference oscillator, two signals are supplied to the synthesizer section from the reference section. The output frequency of the synthesizer section is generated from a Yttrium-iron-garnet (YIG) oscillator which is tuned with a magnetic field. The feedback mechanism that ensures frequency stability is a phase locked loop; however, instead of a fractional-N divide, harmonic sampling is used to divide the output frequency.

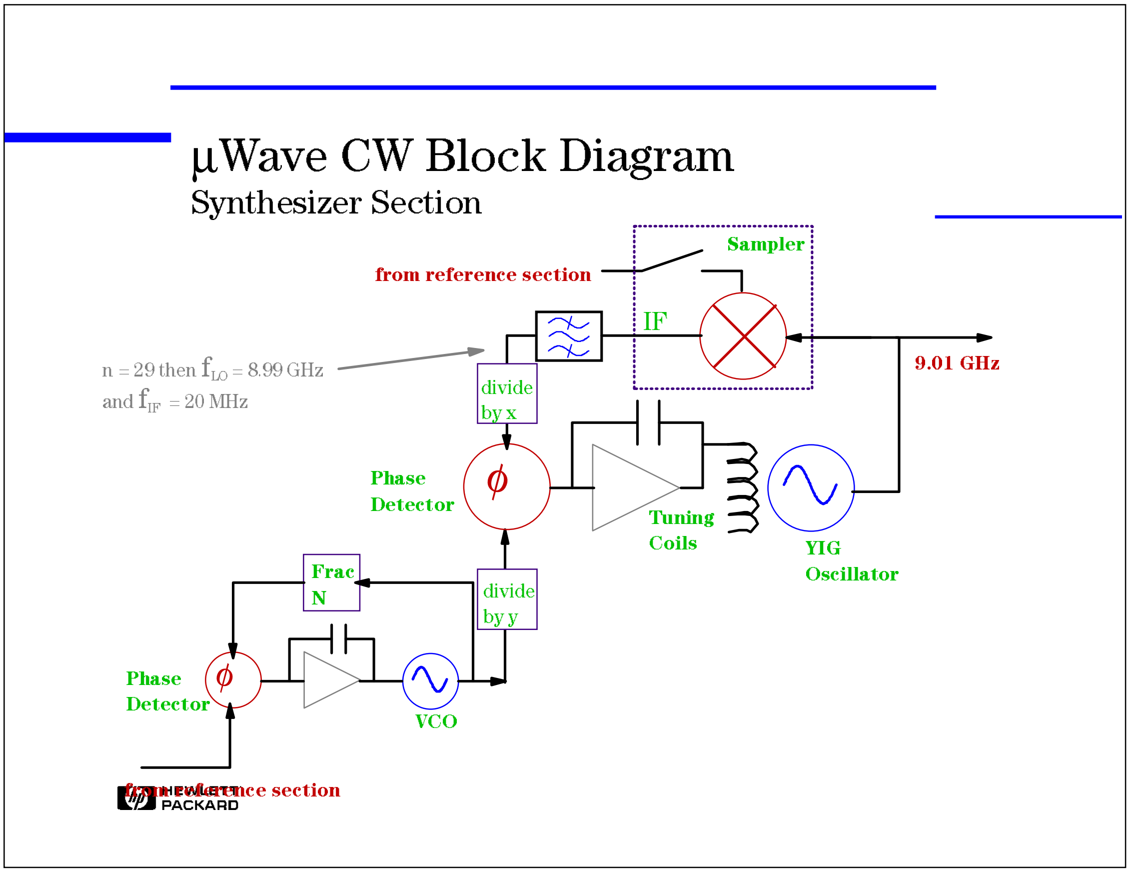
Again, we will take a look at these sections in detail.

Slide #16



The reference oscillator for a microwave source is either a quartz TCXO or OCXO. As with RF sources, the reference oscillator is inside a PLL that may be locked to an external reference. In addition, the reference oscillator drives a PLL that contains an RF VCO. For some microwave sources, this VCO will produce an output frequency in the 300 - 350 MHz range. This higher frequency reference is needed by the harmonic sampler in the synthesizer section.

Slide #17



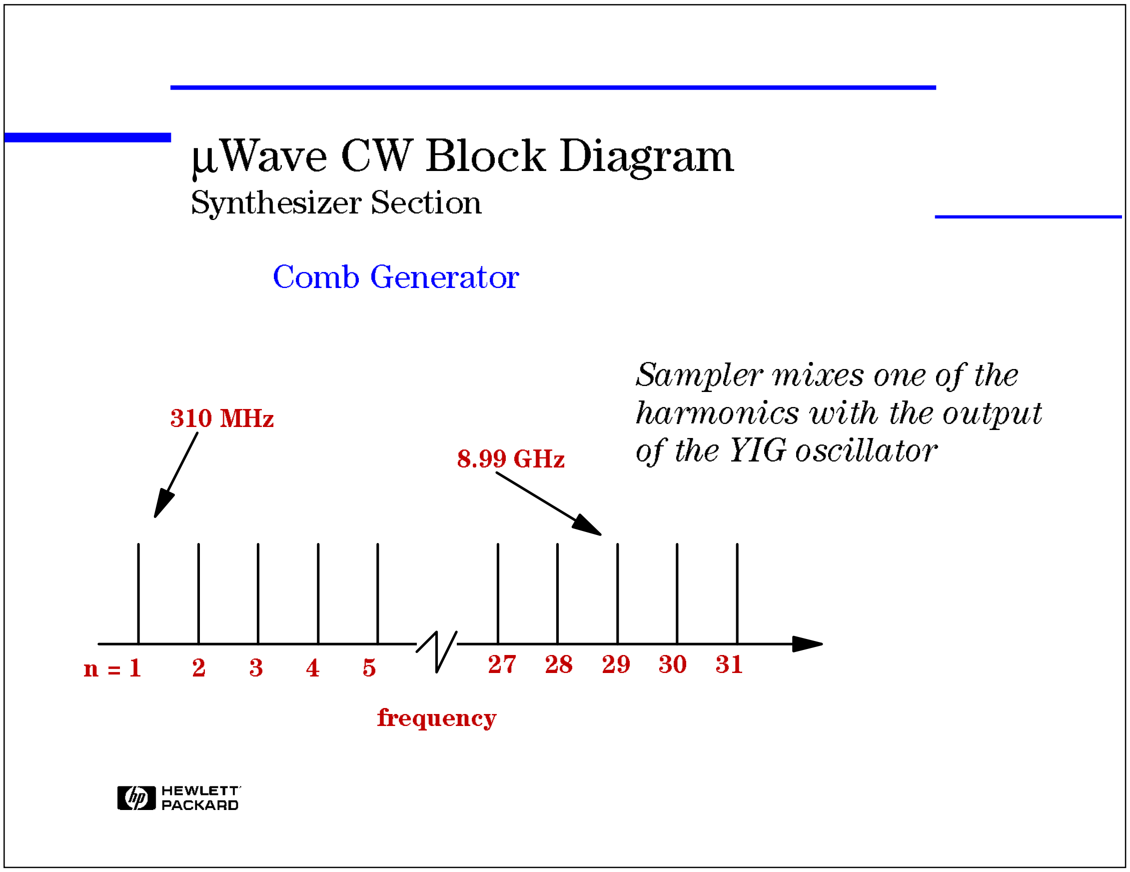
The heart of the synthesizer section is a YIG oscillator. YIG oscillators (YO's) have extremely wide tuning ranges and low phase noise. They are ferromagnetic and are tuned with an electromagnetic tuning coil. In the absence of feedback, YO's do not exhibit good frequency stability.

To achieve good frequency stability, the YO is placed inside a PLL. A portion of the YO's output is sent to the sampler. The reference section supplies the sampler LO. A pulse forming circuit inside the sampler generates harmonics of the LO. Each harmonic mixes with the output of the YO. The IF output of the sampler is filtered. The resulting signal is the YO frequency subtracted from the closest harmonic of the LO. For example, suppose the YO frequency is 9.01 GHz, the sampler LO is 310 MHz, and the IF filter passes signals from 18 MHz to 26 MHz. The YO frequency will mix with the 29th harmonic of the LO, 8.99 GHz, to produce a 20 MHz IF. The 28th harmonic, at 8.68 GHz, will produce an IF output of 330 MHz which will be removed by the bandpass filter. Similarly, the 30th harmonic, at 9.3 GHz, will produce an IF output of 290 MHz which will also be removed.

The reference for the YO PLL is supplied by a VCO. The VCO is stabilized in a separate PLL whose reference is supplied by the reference section.

The output of the phase detector in the YO PLL is used to drive the tuning coils of the YO. This locks the YO to an integer multiple of the sampler LO minus a fixed integer ratio of the YO PLL reference. This technique enables microwave sources to achieve extremely good frequency resolution.

Slide #18



The pulse forming circuit inside the sampler creates a frequency comb. This comb contains the sampler LO and all of its harmonics. The comb enables the YO output to be mixed down to a lower frequency. In place of a sampler, a series of fixed oscillators could be used to mix the YO output down to a level where a fractional-N divide could be used. This adds cost and hardware complexity to the source. Using a sampler does require sophisticated firmware algorithms, however, the cost and complexity of the hardware is reduced.

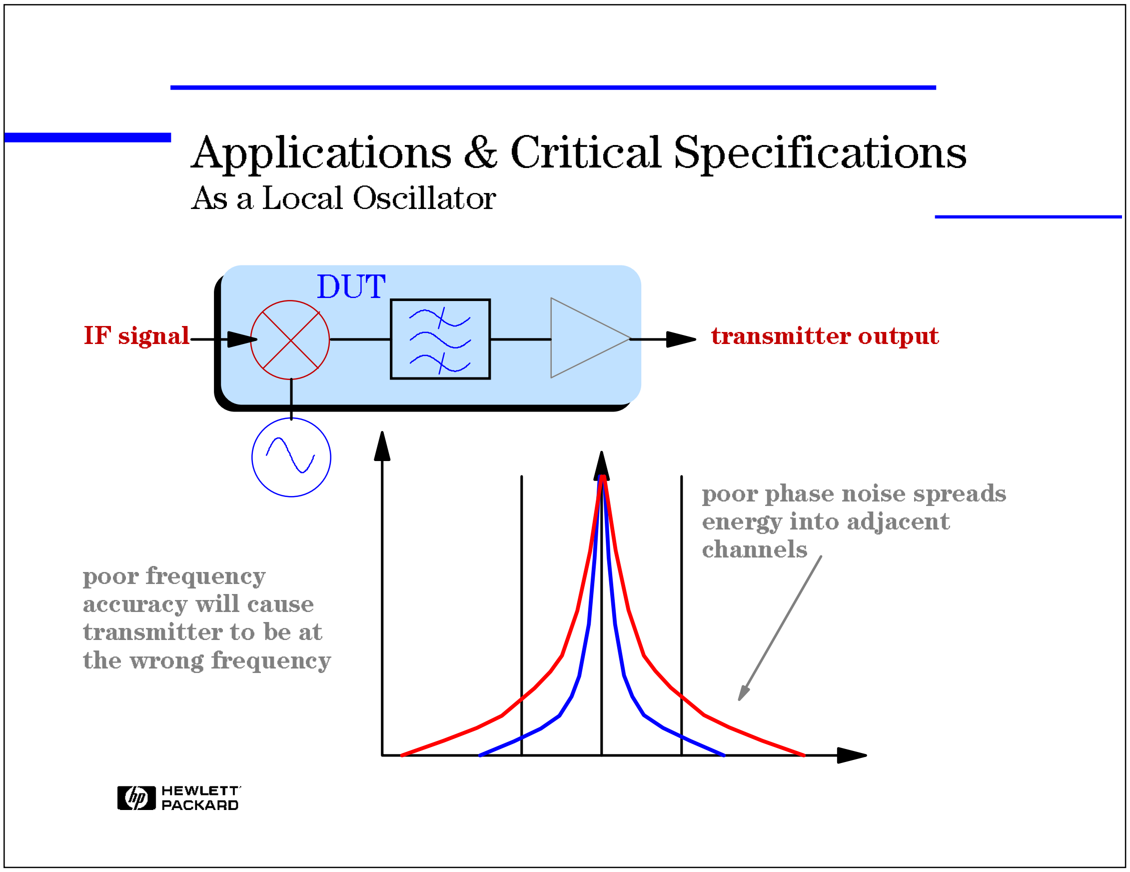
Slide #19

Applications & Critical Specifications

- Local Oscillator
  - phase noise
  - frequency accuracy
- Amplifier Distortion
  - spurious
  - TOI (for system)
- Receiver Testing
  - Spurious
    - spurious
    - level accuracy

There are many, many applications for RF and microwave CW sources. The list above represents only a portion of the more common applications.

Slide #20

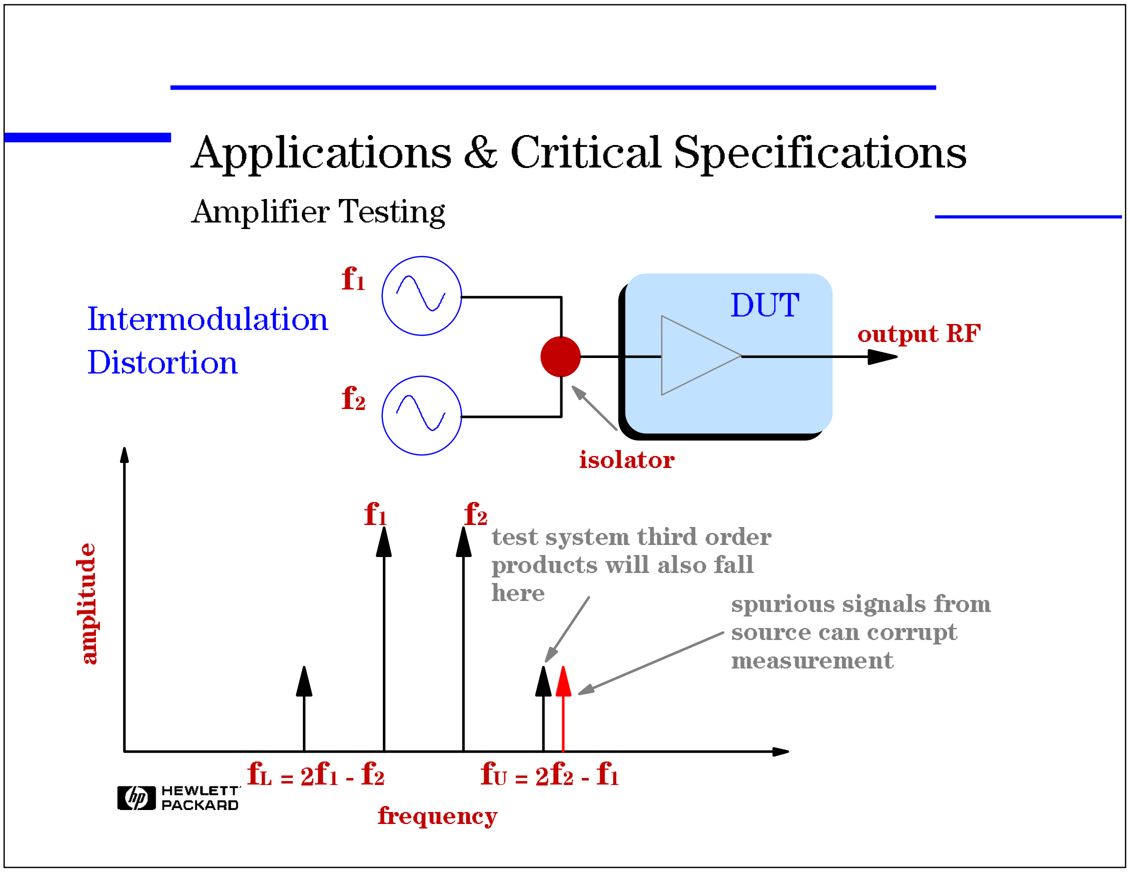


CW sources are often used as local oscillators in the development of transmitters and receivers.

Frequently during development, hardware sections of a prototype become available in stages. CW sources are often used in place of unfinished sections. For example, in the development of transmitter, if all sections except the local oscillator section are finished, a CW source is often used as the local oscillator.

When using a CW source as a local oscillator, phase noise and frequency accuracy are critical. Poor frequency accuracy will, for example, cause a transmitter to transmit at the wrong frequency. In a channelized communication system, poor phase noise will spread energy into adjacent channels. This spread into adjacent channels could be incorrectly attributed to the power amps of a transmitter.

Slide #21



Third order intercept, or TOI, is a common amplifier measurement. In this measurement, two CW sources are combined at the input of an amplifier. The frequencies of each source are slightly offset from each other and yet still inside the bandwidth of the amplifier. The non-linearities of an amplifier will produce third order mixing products:

$$f_L = 2f_1 - f_2$$

$$f_U = 2f_2 - f_1$$

where f_1 and f_2 are the output frequencies of the two sources.

Spurious signals from the CW sources can corrupt the measurement. When selecting an appropriate source, the non-harmonic spurious levels should be well below the third order products produced by the amplifier under test.

The test system can also introduce sources of error. Whenever two signals are input to a combiner, the non-linearities of the sources will create intermodulation products. The intermodulation products, when using a simple combiner, are created by the ALC of the source. The signal from the first source passes through the resistive combiner network and into the second source output with a 6 dB loss and the loss associated with the output attenuation of the second source. Because the two sources are at different frequencies, the sum of the two signals has an AM component equal to the difference frequency. For difference frequencies that are within the bandwidth of the ALC, the ALC of the second source sees this additional power and tries to level the output by adding AM. The intermodulation products created by the test system are at the same frequencies as those created by the amplifier under test.

Intermodulation products can be reduced either through better isolation of the signal sources or by suppressing the power that transfers from one source to the other.

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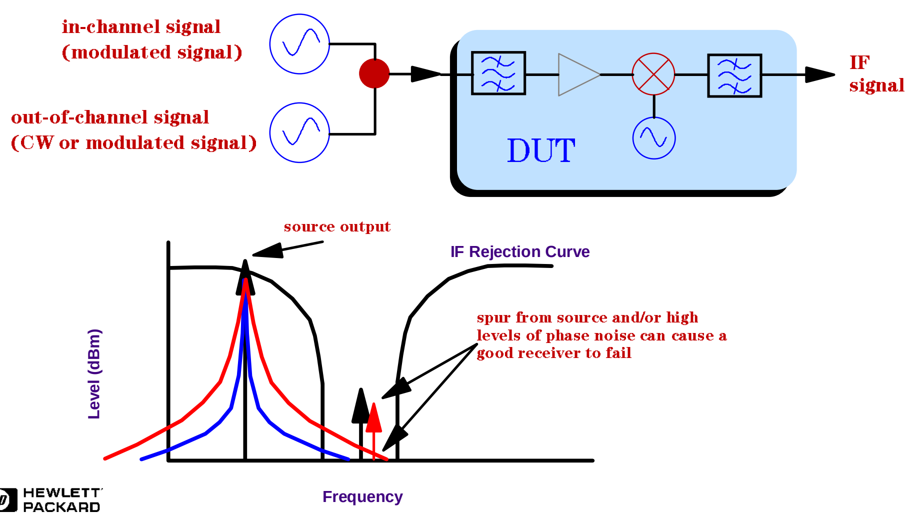
Slide #22

Applications & Critical Specifications

Receiver Testing

Spurious Immunity



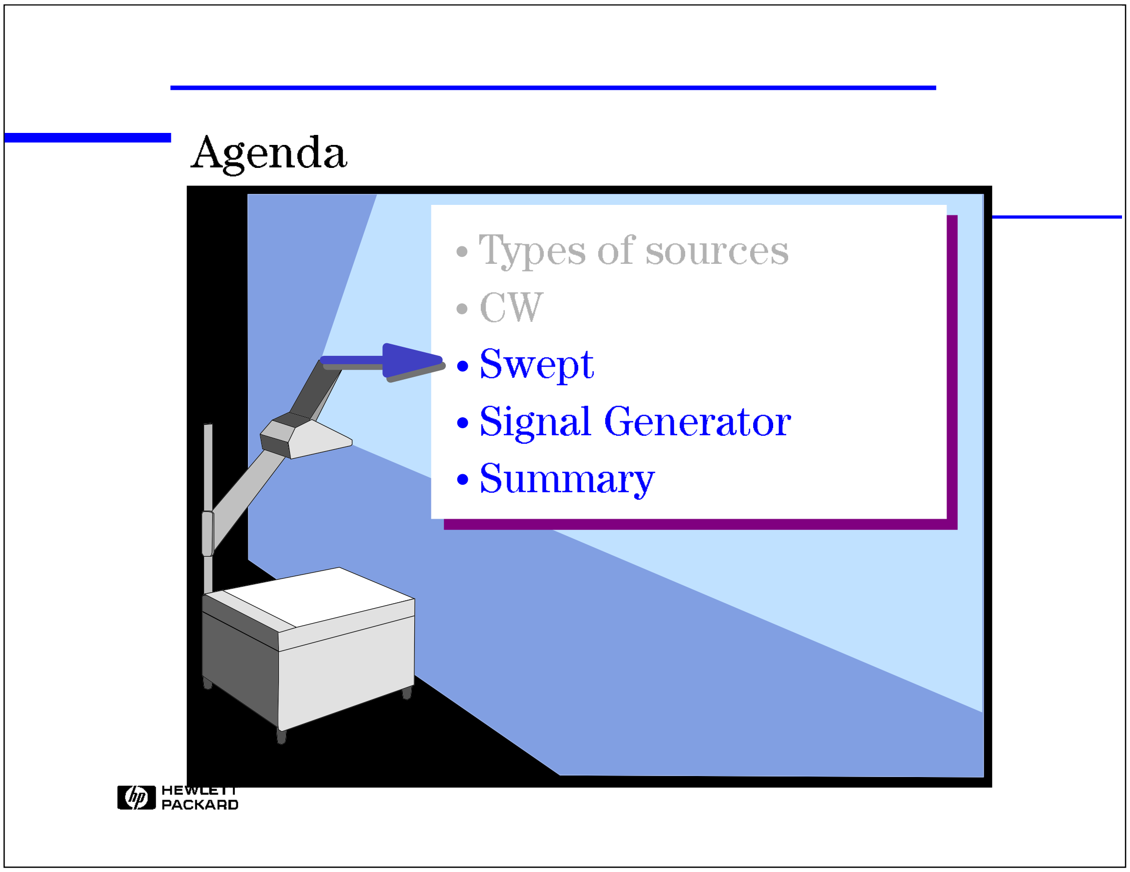
Spurious immunity is a measure of the ability of a receiver to prevent unwanted signals from causing an unwanted response at the output of the receiver. To make this measurement, one source inputs a modulated test signal at the desired channel frequency at a level above the sensitivity of the receiver. The second source outputs an interfering signal over a broad range of frequencies. The interfering signal may be modulated or unmodulated depending upon the frequency range and the communication standard. The output amplitude of the interfering signal is adjusted until the BER (for digital systems) or SINAD (for analog systems) of the receiver under test is degraded to a specified level. The difference between the test signal and the interfering signal is the spurious immunity of the receiver:

$$\alpha_{spur} = P_{interferer} - P_{test}$$

The non-harmonic spurious output of the interfering source must be at sufficiently low levels to ensure that the measurement is not affected. The non-harmonic spurious signals should be about 15 dB below the spurious immunity specification. At this level, spurs from the source will make only a minimal contribution to the overall in-channel noise floor.

The level accuracy also affects the test. An example will illustrate how. If the set level of the interferer is 0dBm and the set level of the test signal is -50dBm then the spurious immunity is 50dB. If the level accuracy of the two sources is +/-1dB, then the level of the interferer might actually be -1dBm and the level of the test signal might actually be -49dBm. This results in an actual spurious immunity of 48dB; not quite as good. By considering the level accuracy of the sources, an uncertainty for the measurement can be determined. In this case, the measured spurious immunity is 50dB +/- 2dB. The uncertainty is ***twice*** the level accuracy.

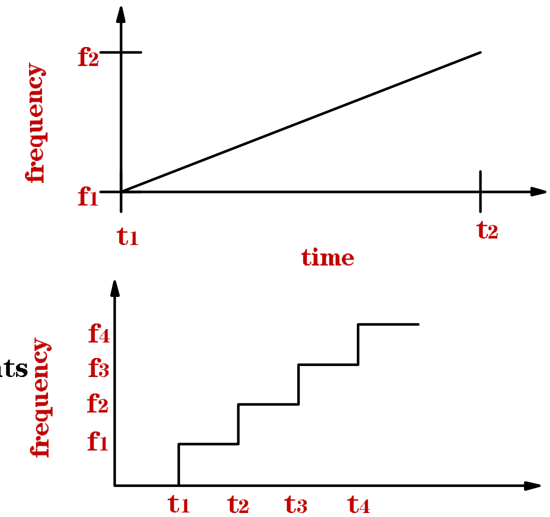
Slide #23


Agenda
• Types of sources
• CW
• Swept
• Signal Generator
• Summary
HEWLETT PACKARD

Slide #24

Sweeper Specifications
...Frequency

- **ramp sweep**
  - accuracy
  - sweep time
  - resolution
- **step sweep**
  - accuracy
  - number of points
  - switching time



Sweepers add the ability to sweep frequency, power, or both. There are two types of frequency sweeps: ramp sweep and step (or arbitrary list) sweep. In ramp sweep, the output sine wave frequency is increased from a start frequency to a stop frequency. This produces a linear frequency versus time plot. In step (or list) sweep, the output frequency is abruptly changed from one frequency to another. The source will then remain at each new frequency for a specified length of time.

For ramp sweep, the accuracy, sweep time, and frequency resolution of the source are usually specified. For step sweep, the accuracy, number of points, and switching time are specified. The number of points may be as few as two or as many as several hundered. The switching time is the time needed by the source to switch from one frequency to another.

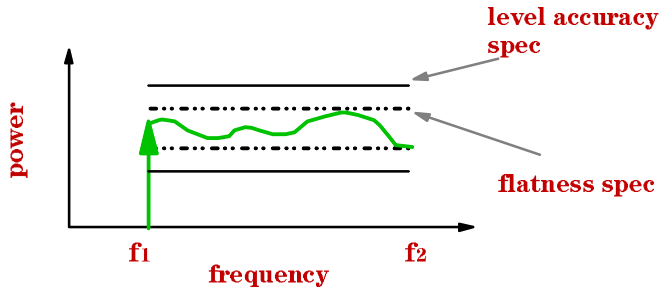
Slide #25

Sweeper Specifications

...Amplitude

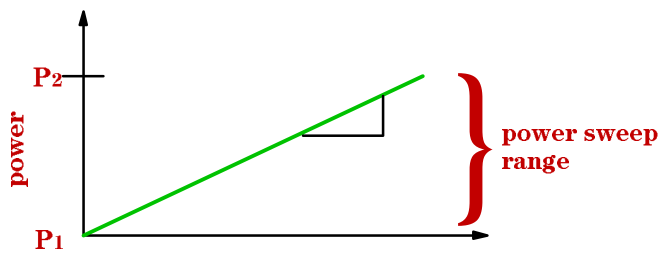
Frequency Sweep

- Level Accuracy
- Flatness



Power Sweep

- Power Sweep Range
- Power Slope Range



The output power will vary by no more than the flatness specification throughout the sweep. In addition, the output power is also constrained to remain within the level accuracy specification of the source. For example, consider a source with a level accuracy of +/- 1.0 dB and a flatness specification of +/- 0.7 dB. If the output is set to 0 dBm, the actual output could really be as high a 1 dBm or as low as -1 dBm. If the actual output is 1 dBm, during the sweep, the power can only drift downward by the 0.7 dB, the flatness specification; the power cannot drift above 1 dBm because the ALC will constrain the power to remain within 1 dB of the set level of 0 dBm.

When sweeping power, the sweep range will determine possible range of output powers. The slope range will determine how quickly the source can sweep from one power to another. In place of a power slope, some sources allow the user to specify the number of points in the power sweep and the dwell time.

Source match is generally specified in standing wave ratio (SWR); SWR is really just a measure of how close the source output is to 50ohms. The value of SWR can range between one and infinity. One is a perfect 50ohm match and infinity is, well, really REALLY bad. If the output of the source is not exactly 50ohms, the SWR value will be greater than one. Some of the power from a source with a SWR greater than one, when connected to a 50ohm load, will be reflected back to the source.

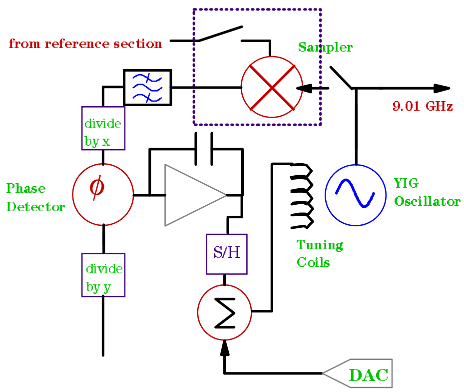
Slide #26

Sweeper Block Diagram

Frequency Sweep: Open Loop

- Phase continuous
- PLL open
- Synthesize start frequencies
- Tuning characteristics must be precisely known

µWave Source:
Synthesizer Section



The sweeper has the same basic block diagram as the CW source. However, some additional hardware is needed to enable the source to sweep the output frequency.

A microwave sweeper, for example, may add a DAC and a summing junction that sweeps the input drive level of the tuning coils. With the YO loop open, the YO output depends solely on the current applied to the tuning coils. When sweeping with the YO loop open, an accurate start frequency is determined with the loop closed. The sample-and-hold block (S/H) holds the proper drive level for the start frequency. An accurate sweep requires that the tuning curve of the YO are precisely known. The tuning curve of a YO plots the output frequency versus the input current applied to the tuning coil. An advantage of sweeping open loop is speed. The major disadvantage of this method is accuracy.

An alternative to sweeping with the loop closed is to sweep with the loop closed. With the loop closed, each frequency point is fully synthesized. This takes longer to sweep but produces a more accurate frequency sweep. There are many microwave and RF sources that sweep closed loop.

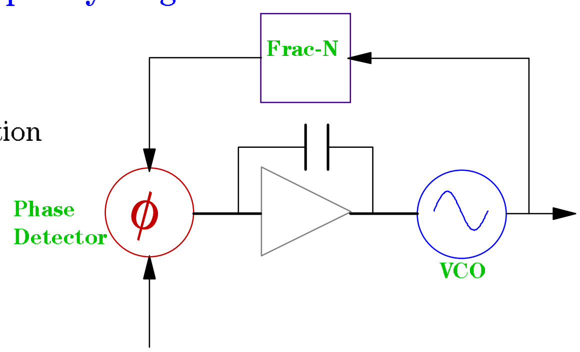
Slide #27

Sweeper Block Diagram

Frequency Sweep: Closed Loop

- Fully synthesized sweep
- Phase continuous within
- PLL never loses lock
- Limited frequency range

RF Source:
Synthesizer Section



The output frequency for a closed loop sweep is fully sythesized throughout the sweep. The PLL never loses lock. This takes a little longer but produces a more accurate sweep.

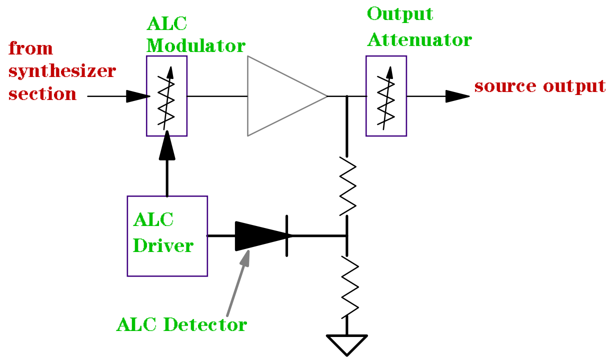
Slide #28

Sweeper Block Diagram

Power Sweep

- Drive ALC Modulator
- Level accuracy maintained
- Broad sweeps may require switching output attentuator

Output Section



Power sweep makes use of the ALC driver and the ALC modulator. A power sweep varies the output power over time. This variation may be linear in which the power is changed by ΔdB per Δt. The power may also be stepped to a pre-determined set of output levels. In either case, the ALC modulator is instructed by the ALC driver to add or subtract power at the appropriate time. Broad power sweeps require switching the output attenuator.

Slide #29

Applications & Critical Specifications

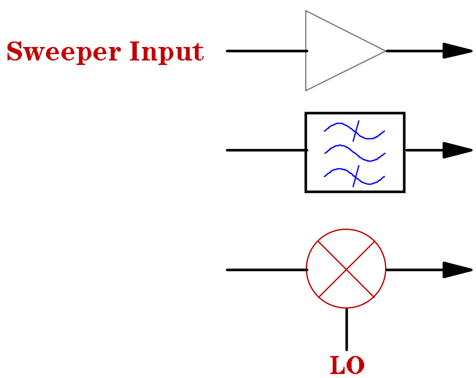
- Frequency Response
  - Frequency Accuracy
  - Output Power (Level) Accuracy
  - Flatness
  - Speed
  - residual FM
- Amplifier Compression
  - Power Range

Frequency sweeps are done to determine the frequency response of devices. Power sweeps, typically done on amplifiers, measure saturation levels.

Slide #30

Applications & Critical Specifications

Frequency Response Testing



- Frequency Accuracy
- Output Power (Level) Accuracy
- Flatness
- Speed
- residual FM

When measuring the frequency response of a device, the following sweeper specifications are important:

Specification	Affect
Frequency Accuracy	center frequency of device under test (DUT)
Output Power (Level) Accuracy	gain or loss
Flatness	flatness
Speed	test cost
residual FM	ability to test high Q devices

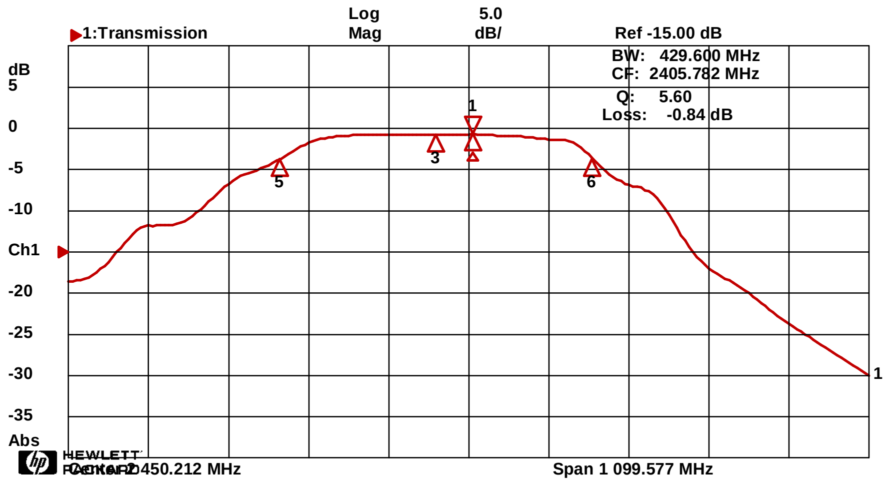
Frequency response measurements are made on many types of devices.

Slide #31

Applications & Critical Specifications

Frequency Response Testing

Who Cares About Accuracy?



Who cares about frequency accuracy? If you're making filters whose 3 dB roll off frequency is better than your competitors, you'd better be able to measure that frequency accurately. A source testing a filter operating near 1 GHz can only set frequency to +/- 10 Hz if the frequency accuracy is 0.01 ppm. Is this accuracy 0.01 ppm enough? Ask your customer: It really depends on the accuracy requirements for parameters such as 3dB bandwidth and gain.

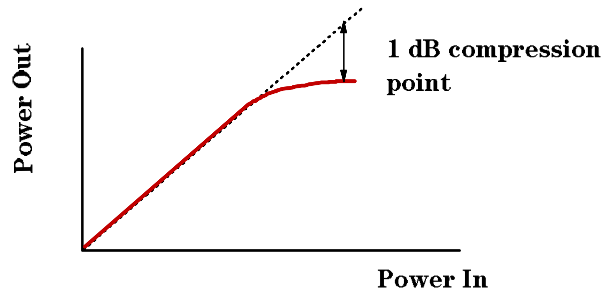
Most channelized frequency communications system employ band pass filters. A frequency inaccuracy can lead to an amplitude measurement error due to the shape of the filter.

Slide #32

Applications & Critical Specifications

Amplifier Compression

- Power Range



The 1 dB compression point is a common amplifier specification used to identify the linear operating range of an amplifier. Power sweep is available on some HP sources.

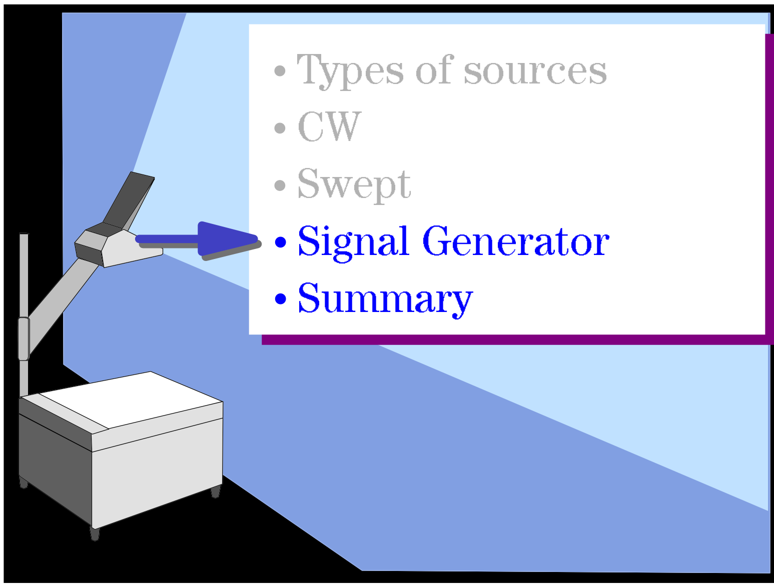
Power sweeps are commonly done on amplifiers to determine 1 dB compression points. A wide power range is needed to drive the amplifier into compression. What cause compression? Consider (again) the output of an amplifier with an input sinewave:

$$
\begin{aligned}
v_o(t) &= a_1 \sin(\omega t) + a_2 \sin^2(\omega t) + a_3 \sin^3(\omega t) + \ldots \\
&= a_2/2 + a_1 \sin(\omega t) + 3a_3/4 \sin(\omega t) + a_2/2 \sin(2\omega t) + a_3/4 \sin(3\omega t) + \ldots
\end{aligned}
$$

When the output no longer tracks the input the amplifier begins to go into saturation. With a high enough input level, the gain of the amplifier is decreased by 1 dB. Where has the power gone? When an amplifier becomes saturated, the output power shifts to the harmonics and heat.

Slide #33

Agenda



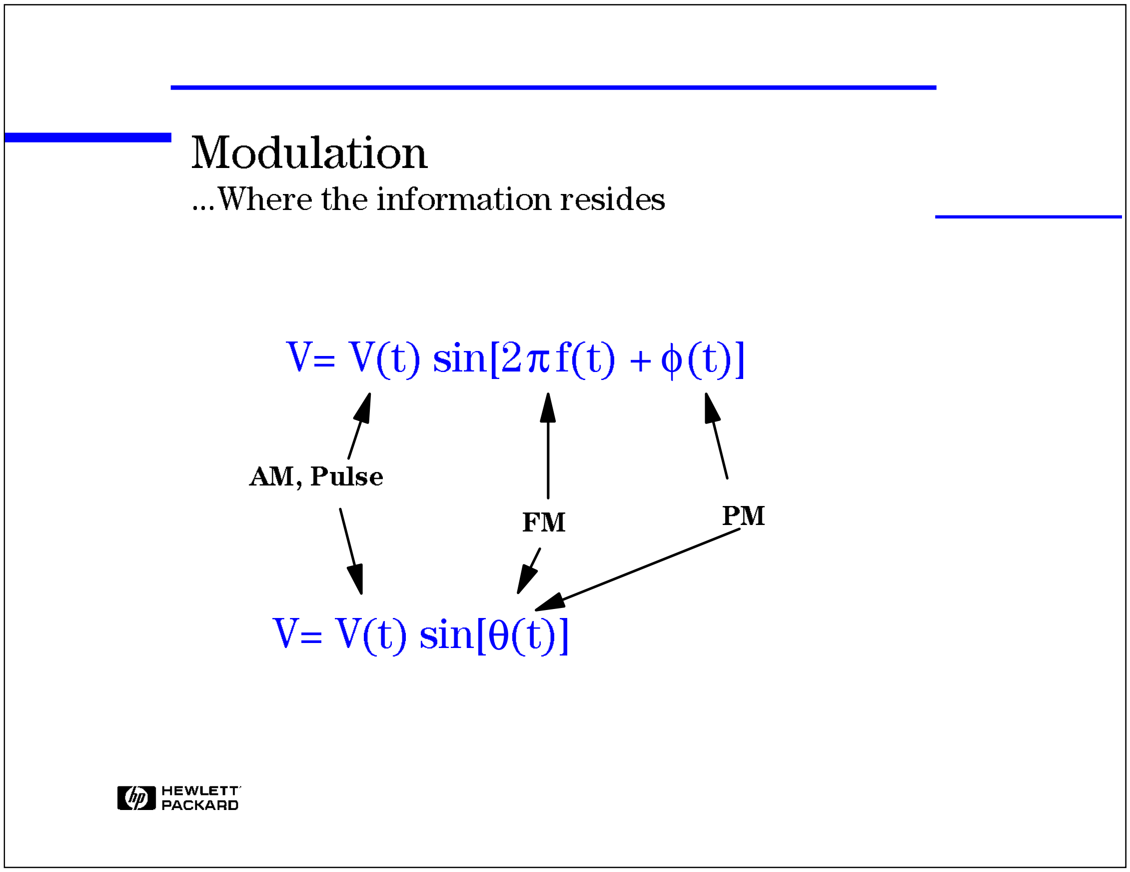
Slide #34

Signal Generators

- Calibrated, variable frequency over a broad range
- Calibrated, variable ouput level over a wide dynamic range
- Calibrated modulation
  - Analog (AM, FM, PM, Pulse)
  - Digital (IQ)
  - Format Specific

A basic signal generator is a source whose output frequency and output level (amplitude) are variable over a wide range and are always known. A signal generator must also have calibrated modulation. The ability to generate modulated signals is the main difference between a signal generator and a CW source.

Slide #35

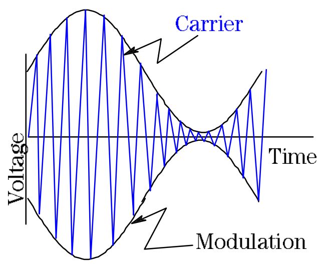


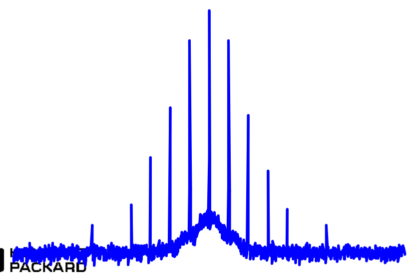
Consider the basic equation of a sine wave. There are three parameters that can be varied: Amplitude, frequency, and phase. Amplitude and pulse modulation are achieved by varying the amplitude of a sine wave. Varying the frequency or phase of the sine wave generates FM and PM. Both FM and PM vary the angle of the sine wave, when viewed in polar coordinates, and may be referred to more generally as angle modulation.

Slide #36

Modulation: Analog

Amplitude Modulation





Important Signal Generator Specs for Amplitude Modulation

- Modulation frequency
- Linear AM
- Log AM
- Depth of modulation (Mod Index)

In amplitude modulation, the modulating signal varies the amplitude of the carrier. The modulating signal carries the information. Amplitude modulation can be represented by the equation:

$$s(t) = A_c \sin(2\pi f_c t)\,[1 + k(t)]$$

where f_c is the carrier and k(t) is the modulation.

Most text book analyses of modulation assume the k(t) modulating signal is a sine wave and, remembering that any waveform may be represented by a sum of sine waves, leaves the more complicated analysis to the student.

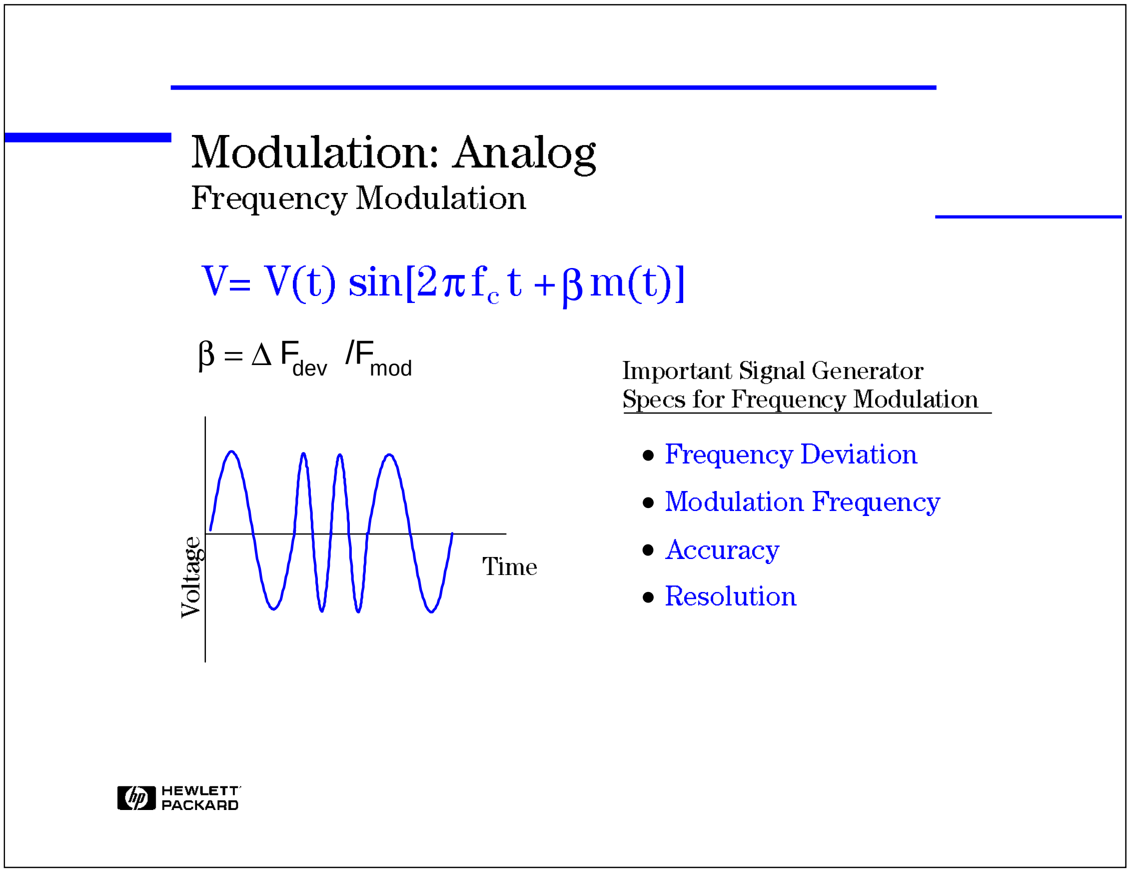
If $k(t) = \mu \sin(2\pi f_m t)$, then:

$$s(t) = A_c \sin(2\pi f_c t)\,[1 + \mu \sin(2\pi f_m t)]$$

This is the classic equation for AM where μ is the depth of modulation, also referred to as the modulation index, and f_m is the modulation frequency. The depth of modulation is defined as the ratio of the peak of the modulating signal to the peak of the carrier signal. When the depth of modulation is expressed as a percentage, the modulation is referred to as linear AM. When the depth of modulation is expressed in "dB", the modulation is referred to as logarithmic AM.

The spectrum of an AM signal contains several sidebands. These sidebands are created from the sums and differences of the carrier frequency and the modulation frequency.

Slide #37



In frequency modulation, the modulating signal changes the frequency of the carrier. The amplitude of the modulating signal determines how far (in frequency) the carrier signal will shift; this is referred to as the frequency deviation or ***ΔFdev***. The frequency of the modulating signal determines how quickly the carrier will shift from one frequency to another; this is referred to as the modulation frequency of ***Fm***.

For FM, the math gets more complicated. For a given frequency deviation and a given rate of frequency change, the modulation index, called **β,** is defined as ***ΔFdev/Fm***.

Frequency modulation, depending on the modulation index, can create an infinite number of sidebands around the carrier. A mathematical solution to frequency modulation requires Bessell functions. The Bessell functions provide an indication of the number and relative strength of the sidebands. The interesting thing about FM is that, with the proper modulation index, the carrier can completely disappear. We will see how this is done later.

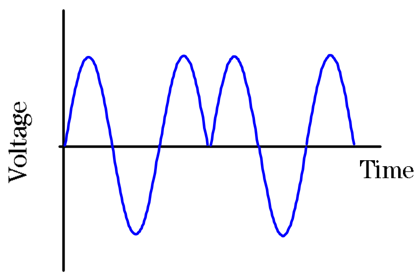
Slide #38

Modulation: Analog

Phase Modulation

$$V = V(t)\sin[2\pi f_c t + \beta m(t)]$$

$$\beta = \Delta\phi_{peak}$$



Important Signal Generator Specs for Phase Modulation

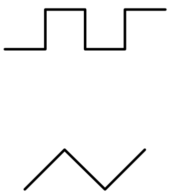
- Phase deviation
- Modulation frequency
- Accuracy
- Resolution

Phase modulation is very similar to frequency modulation. The modulating signal causes the phase of the carrier to shift. The amplitude of the modulating signal determines the phase deviation. The modulation index, β, is defined as the phase deviation of the carrier. Notice that the rate of the phase modulation does not enter into a calculation of β. The spectrum modulation components are spaced as with FM and are determined by the rate of phase modulation, but β will not change if the rate of phase modulation is varied. If β doesn't change, **the shape of the spectrum doesn't change**: only the component spacing changes. This is really the only way of differentiating analog FM from analog PM.

Slide #39

Modulation: Analog

PM is Really the Same as FM...



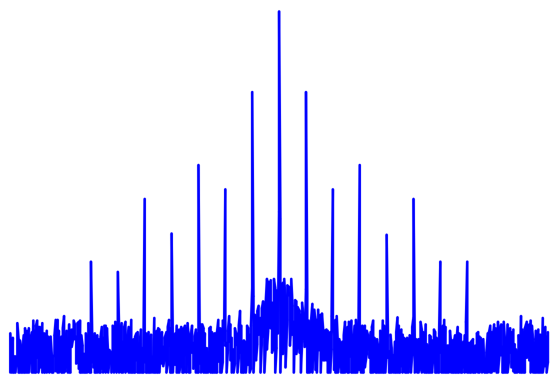
FM Modulator

PM Modulator

$V = V(t)\sin[2\pi f_c t + \beta m(t)]$

$\beta = \Delta\phi_{peak}$

$\beta = \Delta F_{dev} / F_{mod}$



PM is related to FM becasue the rate of change of phase equals frequency (f=dϕ/dt). A phase modulated signal may be generated by either directly varying the phase of a carrier or frequency modulating a carrier with the derivative of the modulating waveform.

An important aspect of both FM and PM is that, ideally, the amplitude of the signal doesn't change during modulation. Because there is no amplitude variation, the linearity requirements of output amplifiers in FM and PM systems is greatly reduced. This is the reason why virtually all wireless digital formats are derived from FM or PM.

Consider the above spectrum. The amplitudes and numbers of the sidebands are determined by solving the modulation equation. Consider the case where the modulation waveform is a sine wave of unit amplitude at frequency fm:

$$V(t) = Vo\sin[2\pi f_c t + \beta\sin(2\pi f_m t)]$$

For β much less than one, the solution to this equation is obtained, using a trigonometric identity, to be:

$$V(t) = Vo\sin[2\pi f_c t\} + Vo(\beta/2)\cos[2\pi(f_c+f_m)t] - Vo(\beta/2)\cos[2\pi(f_c-f_m)t]$$

The general solution results in an infinite series of Bessel functions:

$$V(t) = J_0(\beta)Vo\sin[2\pi f_c t\} + J_1(\beta)Vo\{\cos[2\pi(f_c+f_m)t] - \cos[2\pi(f_c-f_m)t]\} -$$

$$J_2(\beta)Vo\{\cos[2\pi(f_c+2f_m)t] - \cos[2\pi(f_c-2f_m)t]\} +$$

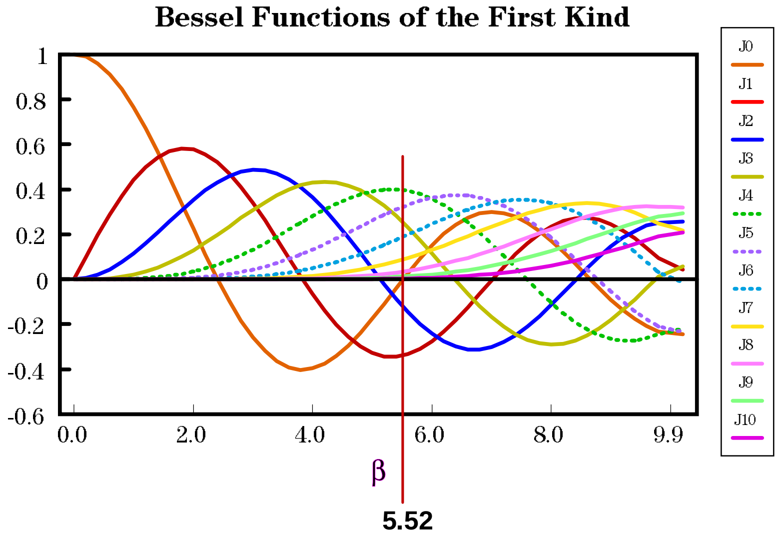
$$J_3(\beta)Vo\{\cos[2\pi(f_c+3f_m)t] - \cos[2\pi(f_c-3f_m)t]\} - \ldots$$

The values of Jn determine the amplitude levels of the sidebands.

Slide #40

Modulation: Analog

Voltages of FM/PM Frequency Components



Earlier the claim was made that for the proper modulation index, the carrier would disappear! Let's see how this happens. Consider an FM or PM spectrum with a β = 5.52. The Bessel functions tell us to expect signals out to eight or nine times the modulation rate. The normalized amplitudes (normalized to the unmodulated carrier amplitude) of these sidebands are (in Volts):

f_0	=	0
$f_{1,-1}$	=	-0.34
$f_{2,-2}$	=	-0.123
$f_{3,-3}$	=	0.251
$f_{4,-4}$	=	0.396
$f_{5,-5}$	=	0.323

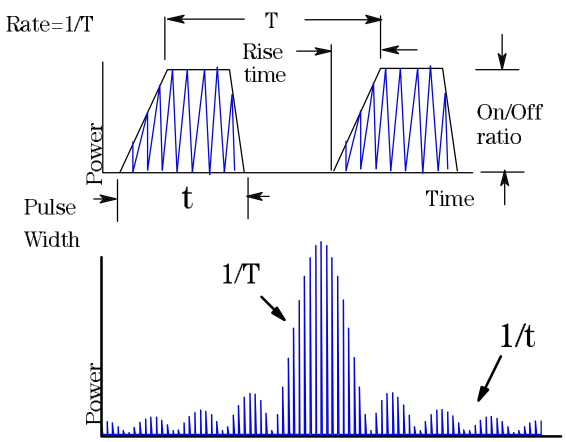
The amplitude of the carrier, f_0, is zero! The FM spectrum is symmetrical in amplitude about the carrier (there are sign inversions): use the positive modulation harmonic to determine the corresponding negative harmonic amplitude value.

Again, these are voltages. Don't forget to square if you're interested in power values.

Slide #41

Modulation: Analog

Pulse Modulation



Important Signal Generator
Specs for Pulse Modulation

- Pulse width
- Pulse period
- On/Off ratio
- Rise time
- Fall time

Pulse modulation is important in both comms and radar applications. In comms, the baseband signal is essentially a pulse and the upconverted signal may be time multiplexed (turned on and off rapidly). A variety of comms, satellite, and radar signals can be generated using a combination of pulse modulation and either FM or PM.

The most important parameters for pulsed RF signals are the pulse rise and fall times, pulse repitition frequency (PRF), pulse period, and pulse width. The line spacings in a pulsed spectrum are separated by the reciprocal of the pulse period. The nulls occur at 1/t where t is the pulse width. The overall shape is a sin(x)/x.

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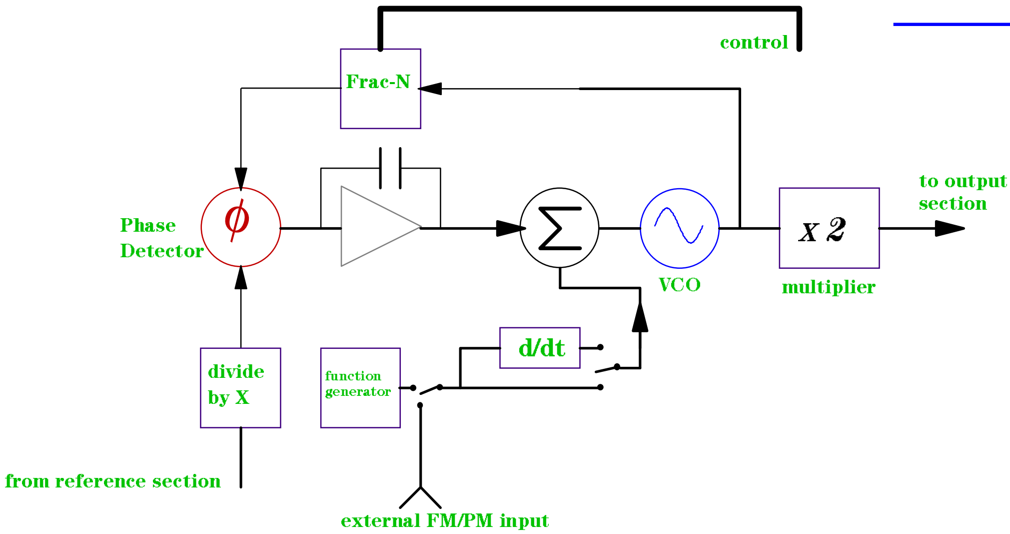
Slide #42

Signal Generator Block Diagram

FM and PM

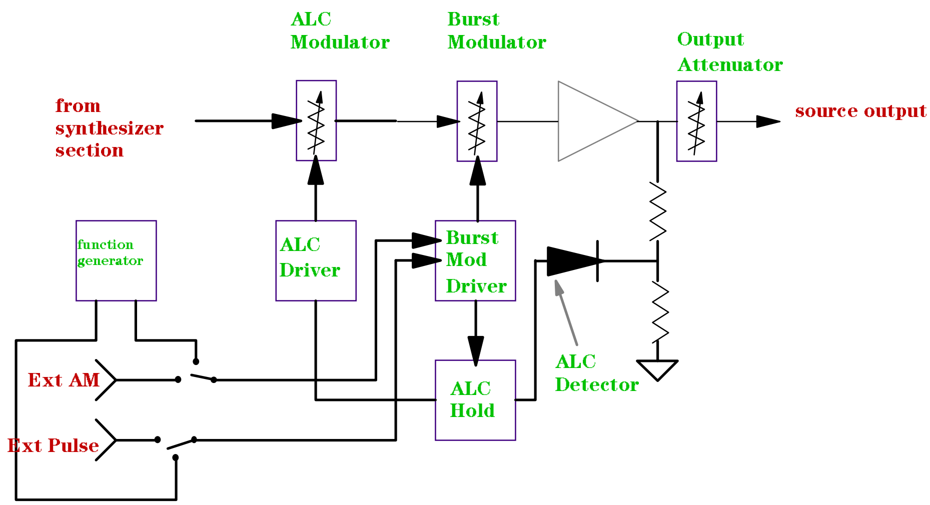


Angle modulation, FM and PM, is produced by directly driving the VCO. The output frequency of the VCO varies with the input voltage. The bandwidth of the PLL limits the maximum frequency or phase deviation that can be achieved by this method. Most signal generators have internal function generators that supply the modulating waveforms. In addition, most signal generators also have external inputs for frequency and phase modulation.

Slide #43

Signal Generator Block Diagram

AM and Pulse

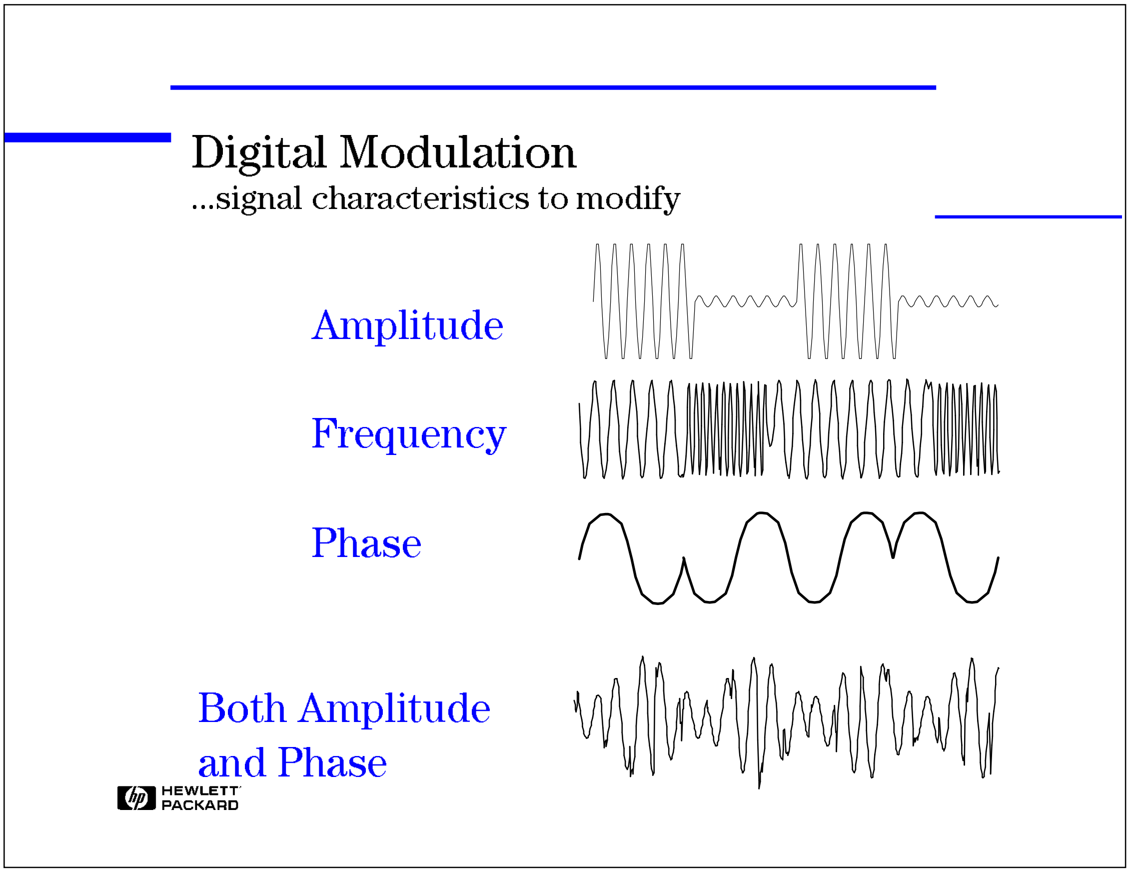


The output section of a signal generator generates amplitude and pulse modulated signals. The **Burst/Modulator Driver** drives a variable attenuator to produce modulated signals. The amplitude of AM and pulsed signals vary as a function of time. The purpose of the ALC is to maintain a constant amplitude. The two operations are in conflict with one another.

For a pulsed signal, when no signal is present (the off time of the pulse) the ALC will add power to the output. This is not desirable. To prevent this, the **ALC Hold** effectively shuts off the ALC circuit when the signal is not present.

For AM signals, the modulation rate is generally greater than the ALC bandwidth; for these signals, the ALC circuit does not respond to the rapid changes in output amplitude. Instead, the output amplitude measured by the ALC detector is effectively averaged over time. For extremely low rate AM signals, however, the ALC circuit will respond by adding and subtracting power to maintain the desired output; the ALC effectively adds additional amplitude modulation. The **ALC Hold** function may be used to shut off the ALC circuit at appropriate times. In addition, the ALC bandwidth may be reduced. In extreme cases, the ALC may be completely shut off.

Slide #44



The only difference between analog (old-fashioned) modulation and digital (new-fangled) modulation is that digital modulation restricts the modulating baseband signal to discrete states rather than allowing the modulating signal to take on any value between a maximum and a minimum value.

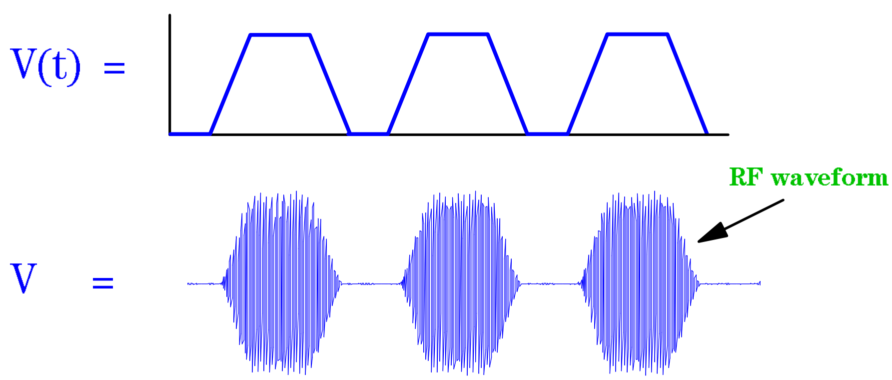
When AM, FM or PM are used in a digital modulation scheme the names become ASK, FSK and PSK. The SK stands for shift keying and is derived from the telegraph key. The modern use implies shifting between discrete states.

Slide #45

Digital Modulation

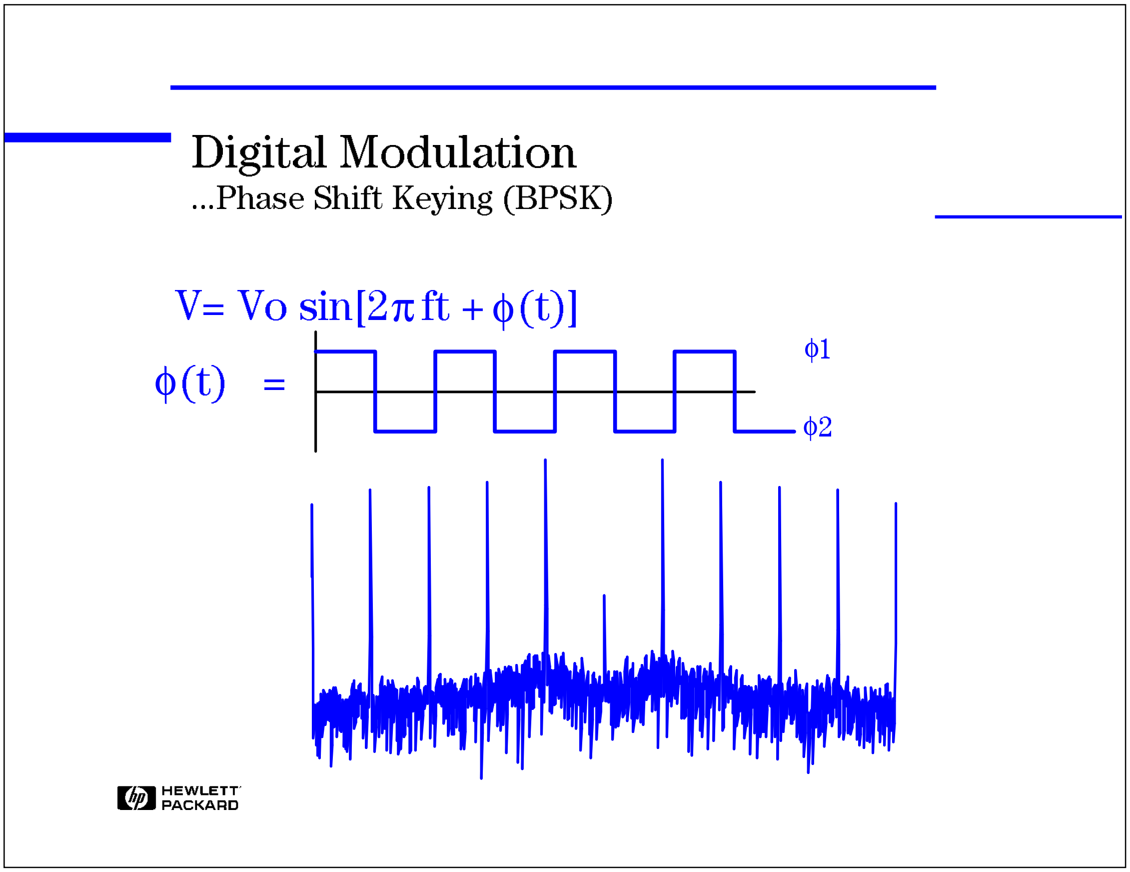
...Amplitude Shift Keying (ASK)

$$V = V(t)\sin[2\pi ft + \phi]$$



Shift keying comes from Morse code which was ASK; the amplitude is turned on and off in amplitude shift keying. Any time you see the phrase "shift keying" as part of a modulation protocol, you know it's digital modulation. The shift keying phrase implies that there are only a limited number of frequency (FSK), phase (PSK) or amplitude (ASK) states allowed. In analog modulation, the change between phase, frequency or amplitude states is continuous.

Slide #46



In phase shift keying, the phase of the carrier signal is shifted between discrete states. There are two common types: Binary phase shift keying (BPSK) and quadrature phase shift keying (QPSK).

BPSK is generated by varying the phase of the carrier between two states that are normally separated by π radians (180 degrees). Let's examine some of the basic properties of BPSK.

BPSK can be achieved by phase modulating a carrier with a square wave. The square wave will force the carrier to change phase between two phase states. Using a square wave causes a very abrupt transition; this creates a very wide spectrum. Most BPSK modulators employ some type of filtering that causes the phase transitions to be less abrupt - this reduces the occupied spectrum of the signal. The BPSK spectrum reflects the discrete nature of the modulation.

Lets look at a slightly more complicated form of phase shift keying, QPSK.

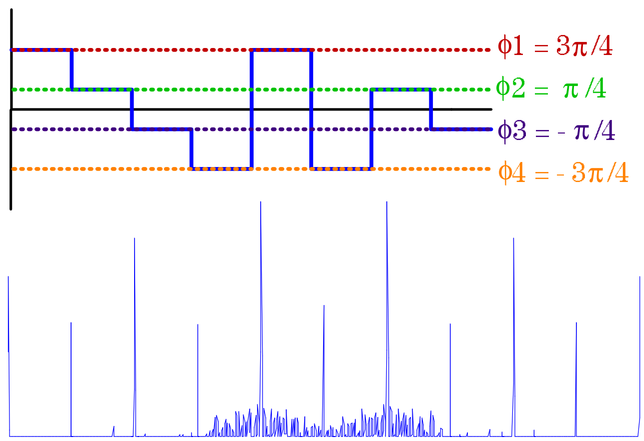
Slide #47

Digital Modulation

...Phase Shift Keying (QPSK)

$$V = V_o \sin[2\pi ft + \phi(t)]$$

$\phi(t)$ =



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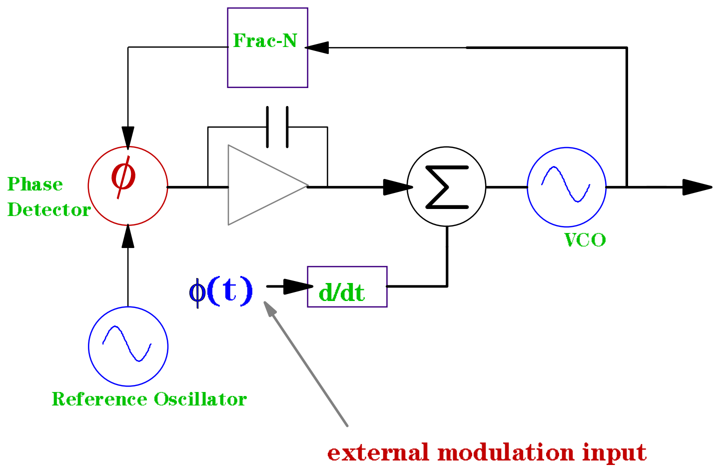
In a QPSK signal, the phase of the carrier is varied between one of four different phase states. These states are normally separated by $\pi/2$ radians. A QPSK signal may be generated by phase modulation with a modulating waveform that contains four discrete levels.

The output spectrum of a QPSK signal is similar to that of a BPSK signal.

Slide #48

Digital Modulation

PSK Implementation: PLL Method



Most signal generators have an external phase modulation input that will support the generation of a QPSK signal. The signal on the preceding slide with the four discrete phase states could be used to create a QPSK signal using the external phase modulation input. When directly phase modulating to generate QPSK, the maximum rate and deviation accuracy of the signal generator limit the performance of the output signal. The maximum rate limits the bit rate of the output. The deviation accuracy limits the achievable modulation quality of a digital signal. There is, however, a better way...

Slide #49

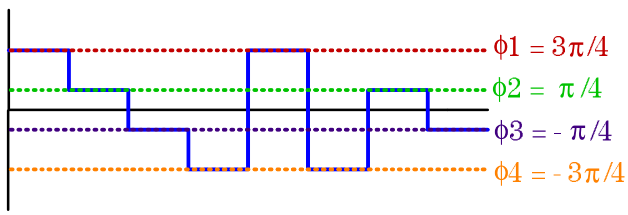
Digital Modulation

PSK Implementation: IQ Method

$$V = Vo \sin[2\pi ft + \phi(t)]$$

$$= Vo \cos[\phi(t)] \sin[2\pi ft] + Vo \sin[\phi(t)] \sin[2\pi ft + \pi/2]$$

$\phi(t)$ =



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Using a simple trigonometric identity, our initial modulation equation may be separated into the sum of two equations. The first equation is a time varying voltage level multiplied by a sine wave. The second equation is a time varying voltage level multiplied by a sine wave that has been shifted by π/2 radians. These two equations are referred to as in-phase (I) and quadrature (Q) components. Using the above expression for ϕ(t), the resulting I and Q signals are:

$$V_I(t) = V_0 \cos[\phi(t)] \sin(2\pi ft) = \begin{pmatrix} \frac{Vo\sqrt{2}}{2} \\ \frac{-Vo\sqrt{2}}{2} \end{pmatrix} \sin(2\pi ft)$$

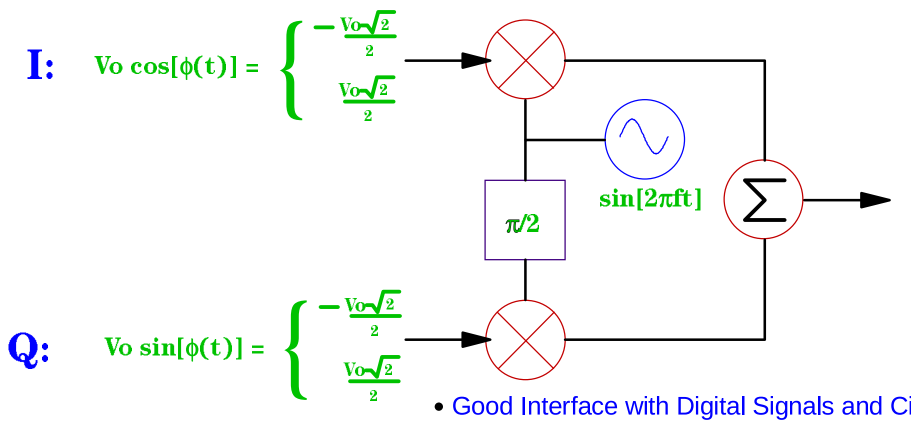
$$V_Q(t) = V \sin[\phi(t)] \sin(2\pi ft + \pi/2) = \begin{pmatrix} \frac{Vo\sqrt{2}}{2} \\ \frac{-Vo\sqrt{2}}{2} \end{pmatrix} \sin(2\pi ft + \pi/2)$$

Both the I and Q signals change discretely with time between two voltage levels. The sum of the two signals represents that same QPSK signal from the previous page.

Slide #50

Digital Modulation

PSK Implementation: IQ Method



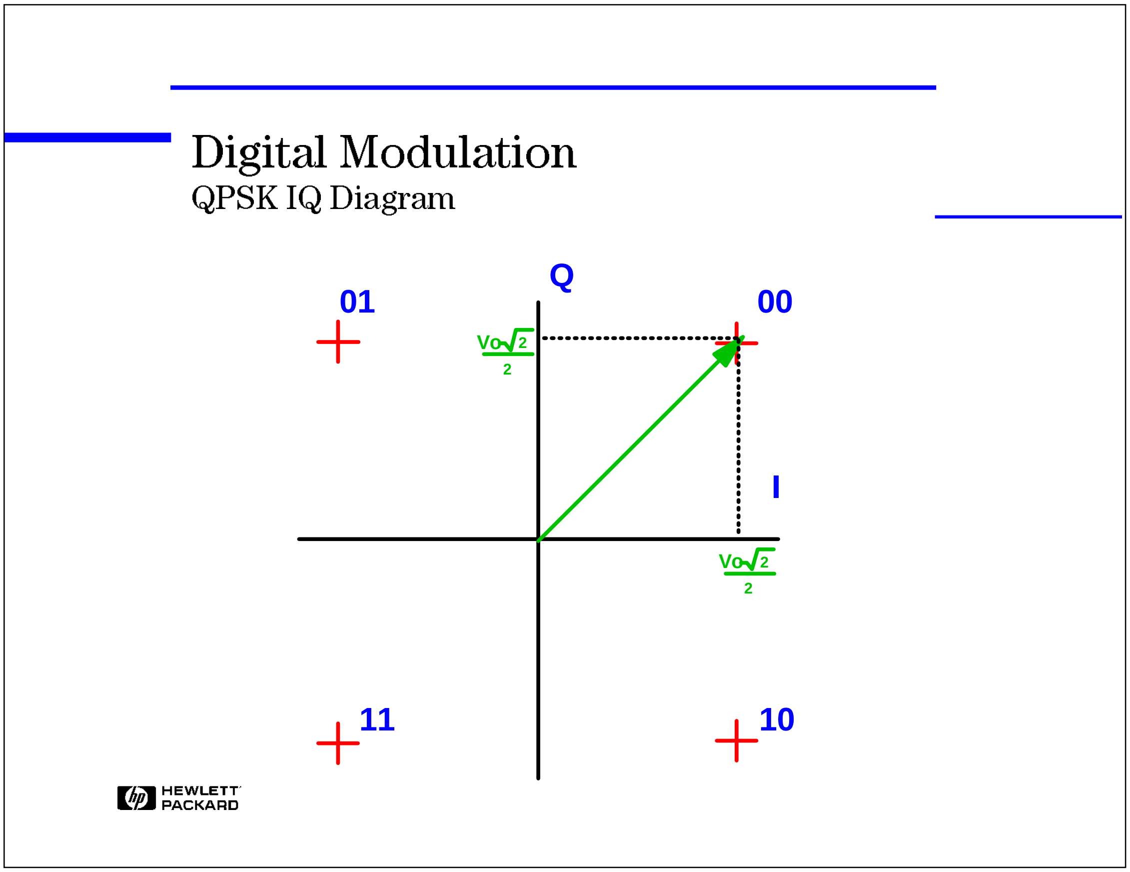
- Good Interface with Digital Signals and Circuits
- Can be Implemented with Simple Circuits
- Can be Modified for Bandwidth Efficiency

Expressing the output signal as the sum of in-phase and quadrature components leads to a natural hardware implementation commonly known as IQ modulation. IQ modulators can be implemented with simple circuits. By modifying the baseband inputs to an IQ modulator, high bandwidth efficiency can be achieved.

IQ modulation is well suited for generating digital signals but may also be used to create traditional AM, FM, and PM signals. When generating a QPSK signal, controlling two voltage states for the I and Q inputs may be done more accurately than changing the phase directly between four different phase states.

Most modern transmitters employ IQ modulation for the generation of digital signals. IQ modulators interface well with digital circuits (e.g. DAC's, DSP processors).

Slide #51



Because they are orthogonal signals, the I and Q signals may be mapped onto a set of orthogonal axes: The IQ plane. A constellation diagram indicates the allowable states. For QPSK, there are four allowable states defined by the set of four IQ points:

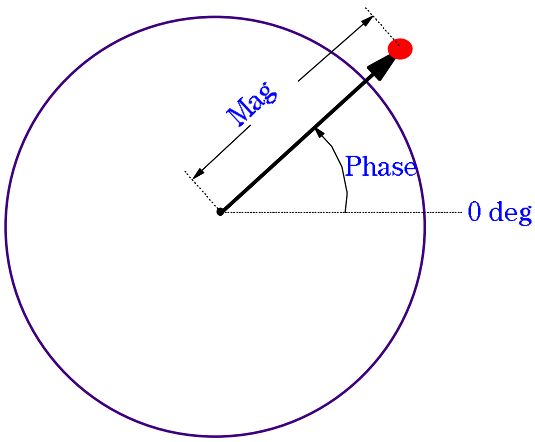
$$\text{IQ:}\quad \begin{pmatrix} \frac{-Vo\sqrt{2}}{2} \\ \frac{Vo\sqrt{2}}{2} \end{pmatrix} \begin{pmatrix} \frac{Vo\sqrt{2}}{2} \\ \frac{Vo\sqrt{2}}{2} \end{pmatrix} \begin{pmatrix} \frac{-Vo\sqrt{2}}{2} \\ \frac{-Vo\sqrt{2}}{2} \end{pmatrix} \begin{pmatrix} \frac{Vo\sqrt{2}}{2} \\ \frac{-Vo\sqrt{2}}{2} \end{pmatrix}$$

The signal at each of these states may be represented by a vector drawn from the origin to the allowable state. The length of the vector indicates the magnitude (power) of the signal. The rotation off of the I axis indicates the phase of the signal (relative to the carrier). The diagram above indicates the position of the signal at one point in time. Over time the vector will rotate from one state to another.

Slide #52

Digital Modulation

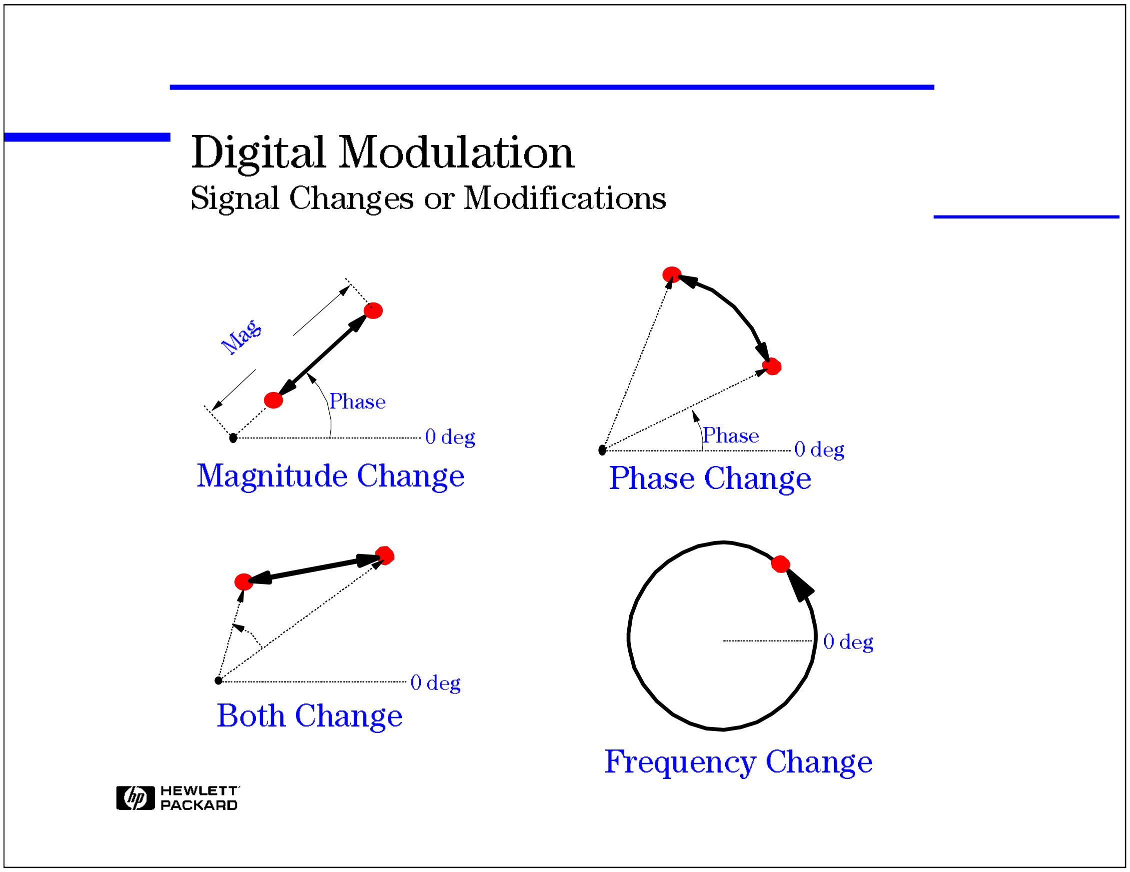
Polar Display: Magnitude & Phase Represented Together



- Magnitude is an absolute value
- Phase is relative to a reference signal

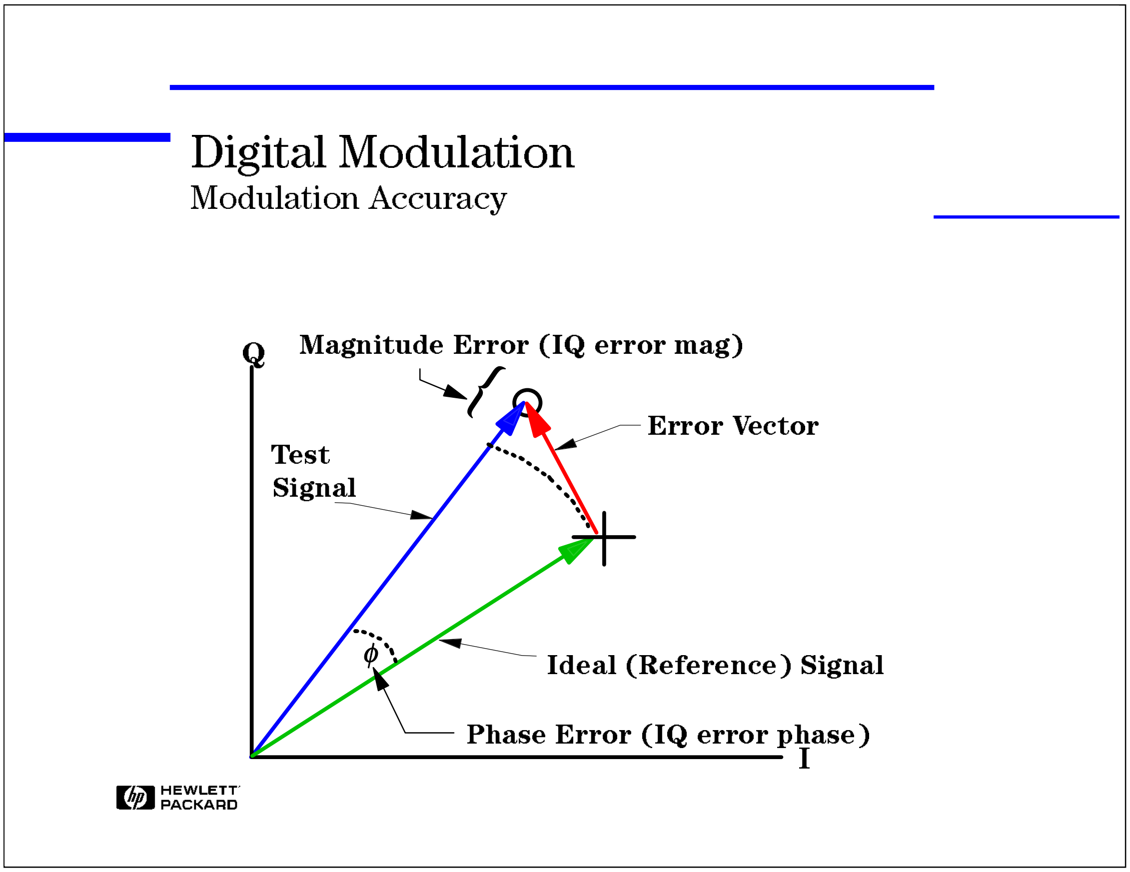
The signal vector, as represented in the IQ plane, is a phasor. The phasor notation provides a convenient way of measuring how the sine wave is changing over time. The phasor doesn't easily provide any frequency information. The rotation of the phasor is referenced to the carrier frequency, therefore the phasor will only rotate if its frequency is different from the carrier frequency.

Slide #53



As mentioned earlier, phasor notation can be used to represent all types of modulation. Amplitude modulation is represented by a magnitude change with no rotation. Phase modulation is represented by a phasor that moves along an arc; the length of the arc indicates the maximum phase deviation. Simultaneous amplitude and phase modulation is indicated by a phasor whose length and phase change with time. Frequency modulation results in a phasor that rotates clockwise or counterclockwise.

Slide #54

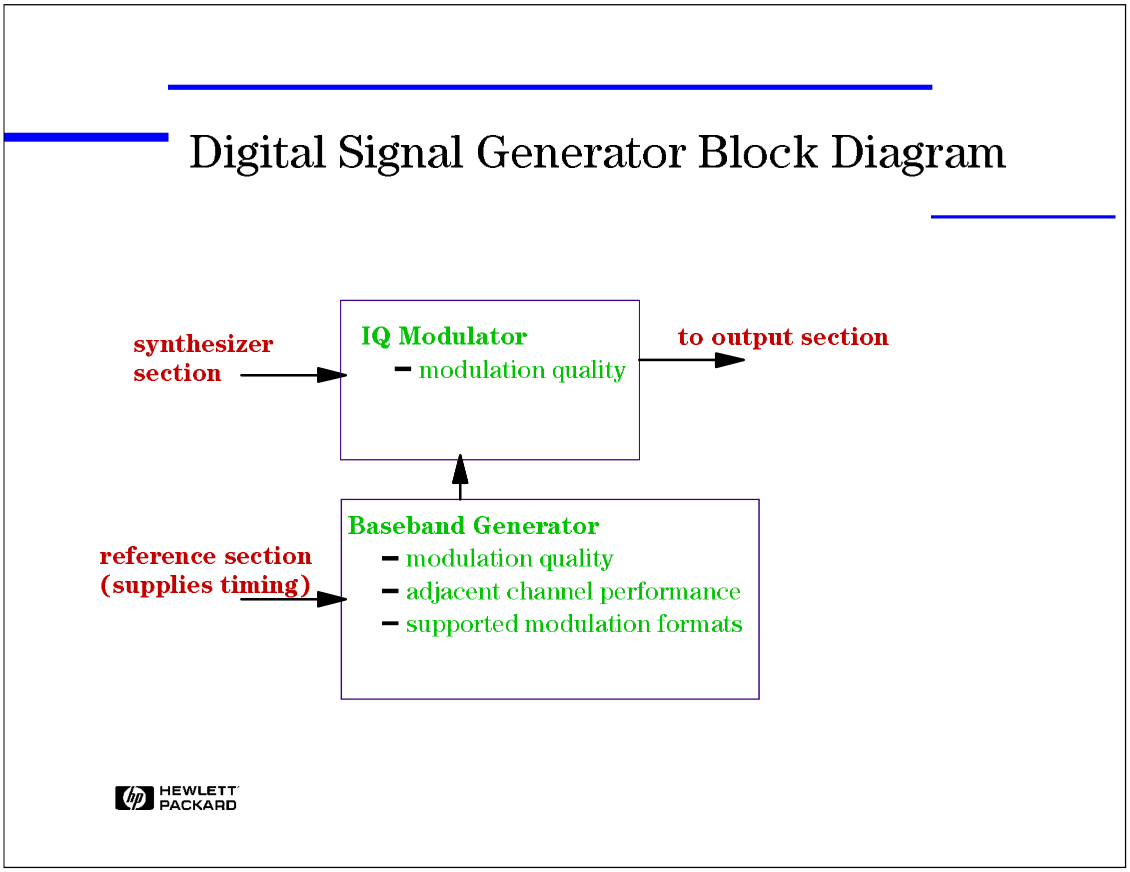


The IQ constellation of our digitally modulated signal provides a wealth of information. Recall first the basics of digital modulation: Digital bits are transferred onto an RF carrier by varying the carrier's magnitude and phase such that, at each data clock transition, the carrier occupies any one of several unique phase and amplitude locations on the IQ plane. Each location encodes a specific data symbol, which consists of one or more data bits. A constellation diagram shows the valid locations at the decision time (i.e., the magnitude and phase relative to the carrier) for all permitted symbols, of which there must be 2^n, given n bits transmitted per symbol. Thus, to demodulate the incoming data, one must accurately determine the exact magnitude and phase of the received signal for each clock transition.

At any moment in time, the signal's magnitude and phase can be measured. These values define the actual or "measured" phasor. At the same time, a corresponding ideal or "reference" phasor can be calculated, given knowledge of the transmitted data stream, the symbol clock timing, baseband filtering parameters, etc. The differences between these two phasors provides both the signal error vector magnitude (EVM) and the phase error. By convention, EVM is reported as a percentage of the ideal peak signal level, usually defined by the constellation's corner states.

EVM and phase error are the two principal parameters for evaluating the quality of a digitally modulated signal. A typical source EVM is around one percent.

Slide #55

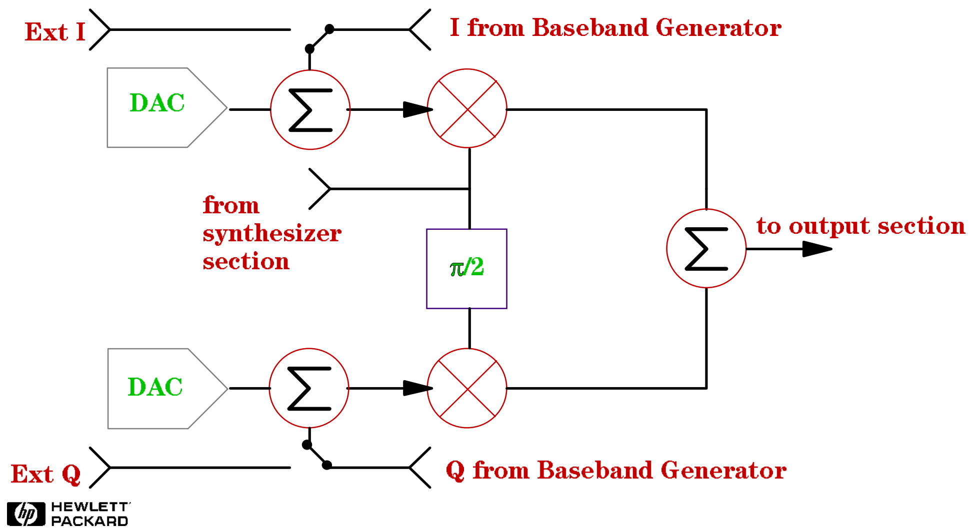


A digital signal generator is created by adding two new blocks to the basic block diagram of a signal generator: an IQ modulator and a baseband generator.

Slide #56

Digital Signal Generator Block Diagram

IQ Modulator



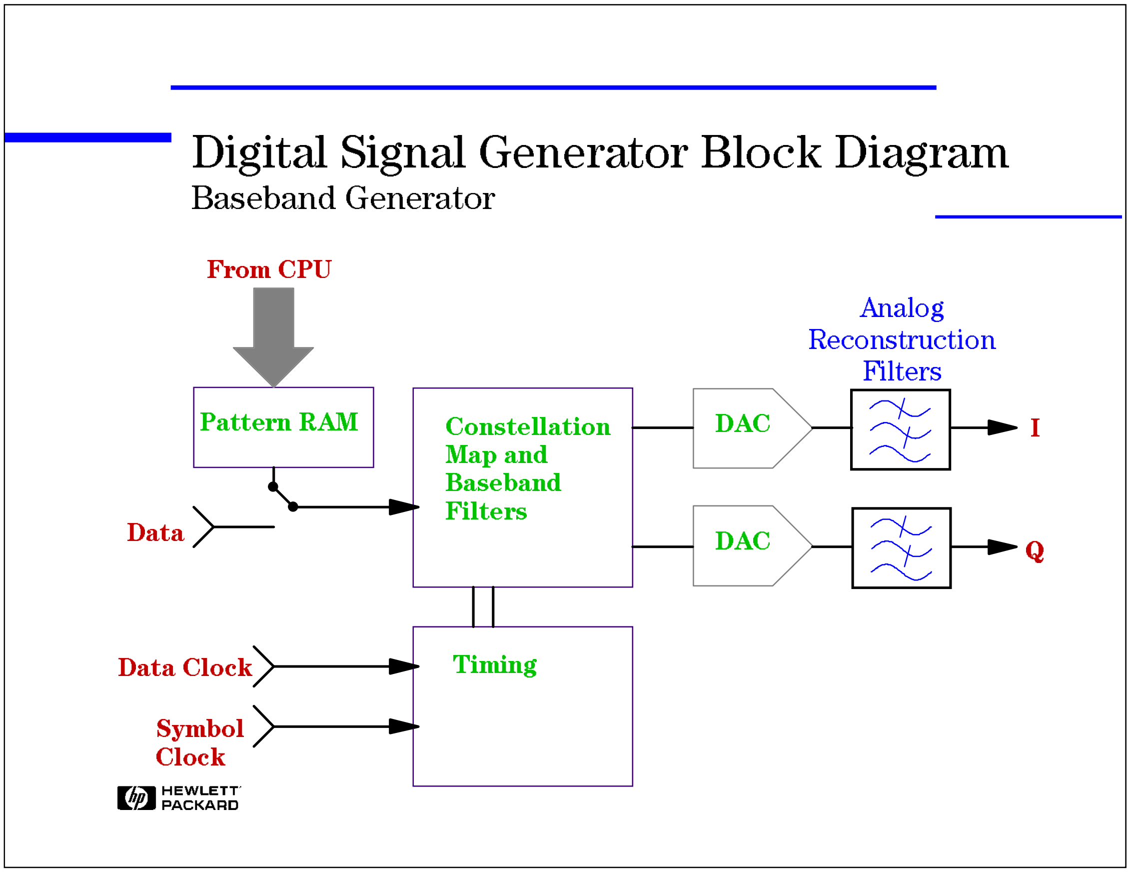
The baseband information for the IQ modulator section can come from two paths: the internal baseband generator or the external I and Q inputs. The information is summed with calibration factors that are stored in ROM look-up tables and are output by a pair of DAC's. This enables the modulator to provide a calibrated, high quality output over a broad range of frequencies.

Today most wireless communications transmitters use an IQ modulator. External I and Q inputs enable users to create custom signals or to test their own baseband generators. When using external sources for I and Q, the input drive level is important. Most IQ modulators have an optimum drive level for maximizing the output signal quality:

$$\overline{V_{input}(t)} = \sqrt{\overline{I^2_{rms}(t)} + \overline{Q^2_{rms}(t)}}$$

For I and Q waveforms that have large peak to average ratios, the input drive level needs to be reduced below the optimum level. This results from the fact that at large peak to average ratios, the peak signals can cause a significant amount of spectral regrowth at the output of the signal generator.

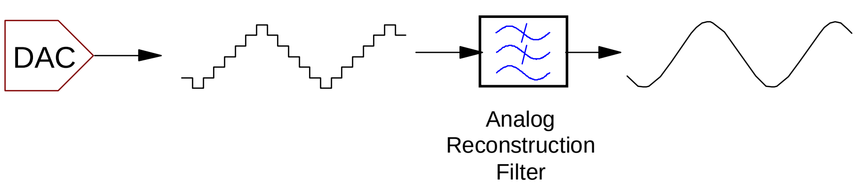
Slide #57



The baseband generator creates the baseband waveforms needed to drive the IQ modulator. The above block diagram provides three paths for supplying data for the baseband waveforms:

1. Data may be loaded from an external computer
2. Data may be loaded directly from RAM
3. TTL data may be input (this requires both a data clock and symbol clock)

The data is the raw "1's" and "0's" that will be used to construct the baseband signal. Datum are combined to make symbols for modulation formats that require more than one bit per symbol. Filtering is added to increase the bandwidth efficiency of the output signal. The filtering is done digitally. The digital stream is sent to a DAC. The output of the DAC's are sent to analog reconstruction filters. These filters are smoothing filters that remove the high frequency components of the waveforms that cause spectral spreading:



Slide #58

Digital Signal Generator

Digital Signals

US-TDMA	
Parameter	**Specification**
Access Method	TDMA/FDD
Modulation	π/4 DQPSK
Channel Bandwidth	30 kHz
Reverse Channel Frequency Band	824 - 849 MHz
Forward Channel Frequency Band	869 - 894 MHz
Filtering	0.35 RRC

The table above illustrates the types of parameters that are specified for digital formats.

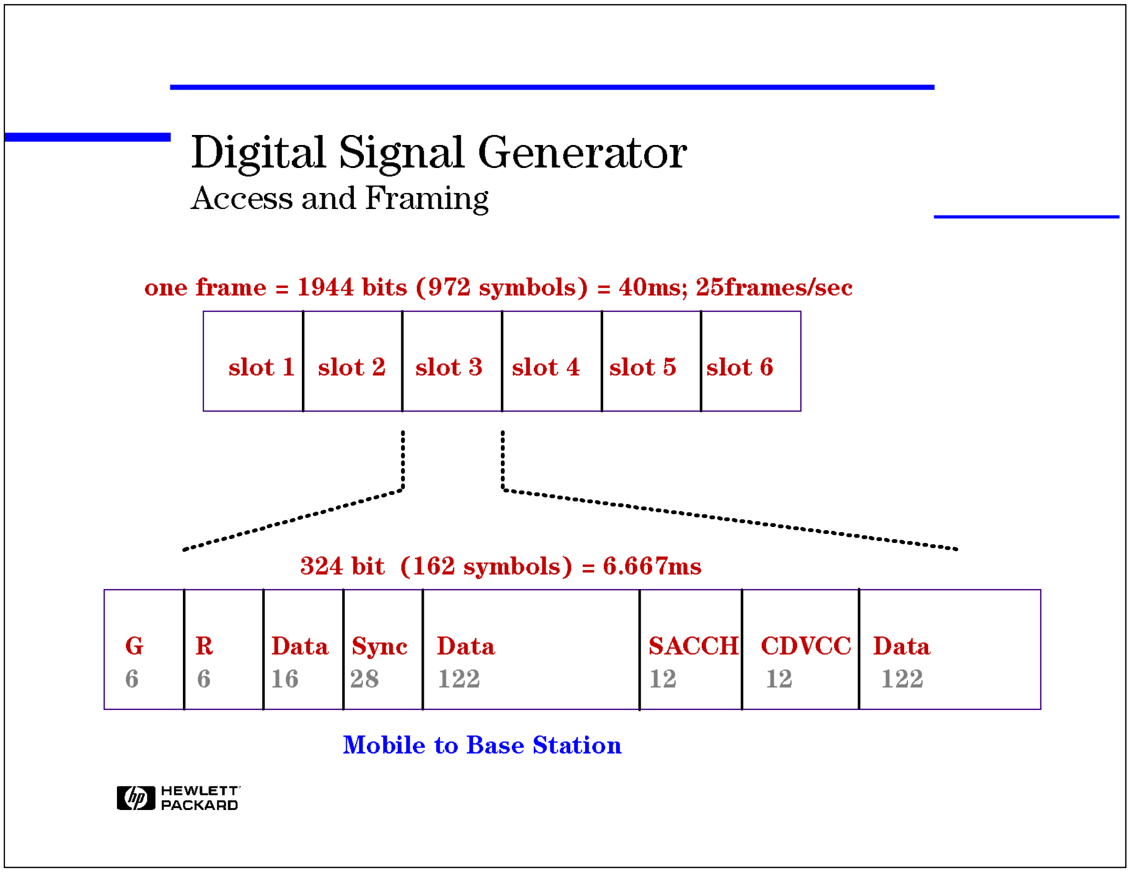
The combination of the baseband generator and the IQ modulator produce a digitally modulated signal. The filtering and modulation type are determined by the shape of the baseband waveforms.

To accurately simulate a digital signal, however, the signal generator must do more than just output the proper modulation. Most digital communication formats, in an effort to conserve bandwidth, have some access scheme. For US-TDMA (IS-54), the access scheme requires a separation in frequency between the forward, base to mobile, and reverse, mobile to base channels. In addition, the forward and reverse links are each allocated a specific slice of time during which communication takes place: This is called **T**ime **D**ivision **M**ultiple **A**ccess or TDMA. This "slice of time" is generally referred to as a time slot.

The following formats are commonly found in digital signal generators:

1. GSM
2. DECT
3. Tetra
4. US-TDMA (NADC, USDC, or IS-54)
5. PDC
6. PHS
7. PHP
8. CDMA (IS-95, wideband)

Slide #59



For US-TDMA, the time slot is part of a frame; there are six time slots per frame. GSM, another TDMA format, uses eight time slots per frame and combines frames into multiframes and superframes.

The bits in a timeslot are grouped to perform various functions. For example, in a US-TDMA system the SACCH and CDVCC are special groupings of bits:

1. CDVC C (Coded Digital Verification Color Code): Handshaking between base and mobile
2. SACCH (Slow Associated Control Channel): Communicate power level changes, hand-off requests

The transmitted voice is contained in the data fields. During testing, standard and special bit patterns are placed in these data fields. Psuedo-random sequences, such as PN9 and PN15 sequences, are examples of standard bit patterns. Special sequences are generally defined by the user.

Signals that are representative of "real world" signals are needed to fully test receivers. Sophisticated signal generators provide the capability to produce signals that are properly modulated, emulate different access schemes (e.g TDMA), and incorporate some level of protocol.

Slide #60

Applications and Critical Specifications

Analog and Digital

- **Receiver Sensitivity**
  - frequency accuracy
  - level accuracy
  - error vector magnitude
- **Receiver Selectivity**
  - phase noise
  - spurious
  - spectral accuracy
- **Spectral Regrowth**
  - ACP performance

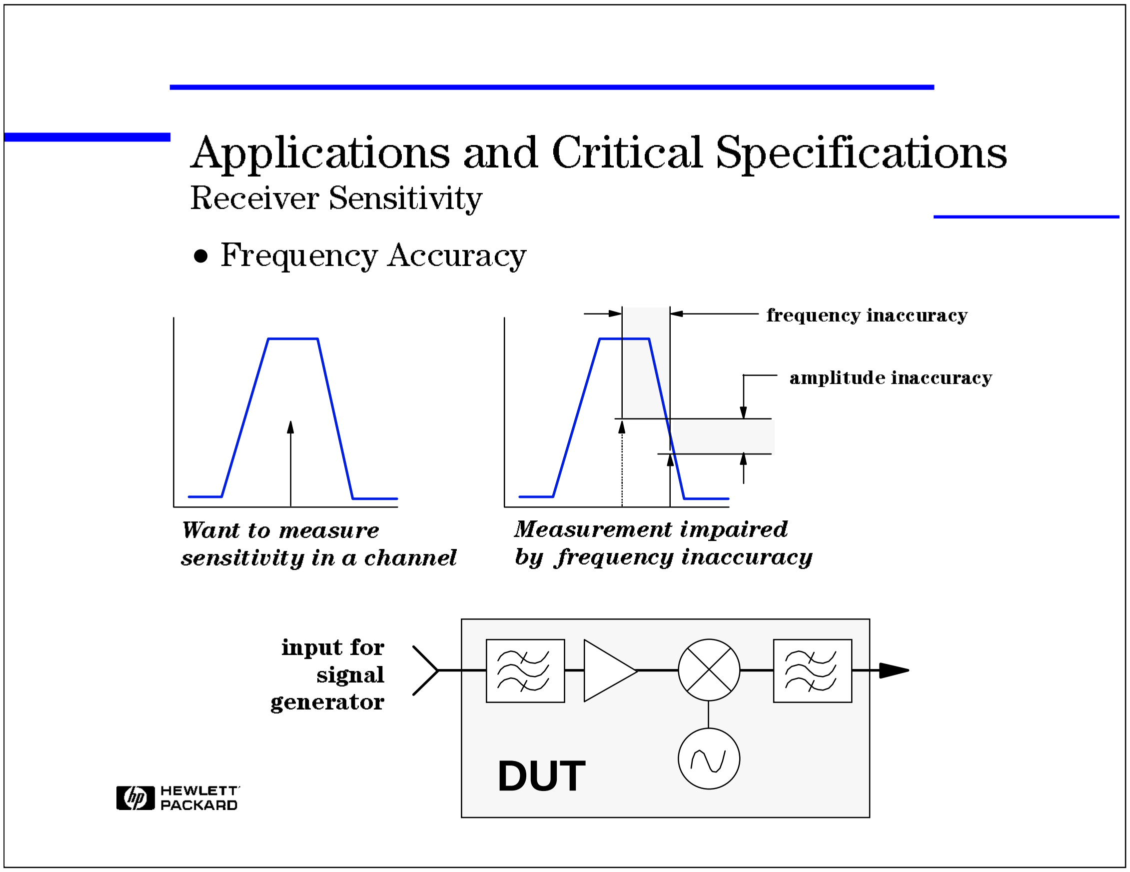
An entire seminar could be devoted to applications of signal generators. Signal generators are used to test receivers as well as the components in a receiver. Sensitivity and selectivity are two receiver tests that are required by most standards. In addition to these two, other common receiver tests include:

1. Co-channel immunity
2. Noise figure
3. Intermodulation rejection

Most standards include detailed descriptions of how these tests are performed. Spectral regrowth is a common transmitter and amplifier test.

When making any test, the specifications of the signal source must be analyzed to ensure that the source does not corrupt the measurement. The critical specifications that a source must meet vary depending upon the test.

Slide #61



The sensitivity of a receiver is the lowest possible signal level that can be reliably detected. Sensitivity is one of the key specifications for a receiver and is generally specified at a particular SINAD for FM receivers or BER for receivers of digitally modulated signals. For FM receivers, SINAD is a figure of merit used to describe the usable signal out of a receiver. SINAD is the ratio of the signal plus noise plus distortion to the noise plus distortion at the same output:

$$\text{SINAD} = 10 \log\left(\frac{S+N+D}{N+D}\right)$$

The level of RF input required to maintain a SINAD of 12 dB is generally defined as the sensitivity of the receiver because this level provides a good quality audio signal. For receivers of digitally modulated signals, sensitivity is defined as the level of the received signal that produces a specified BER when the signal is modulated with a specified psuedo-random binary sequence (PRBS) of data.

The following critical signal generator specifications for measuring the sensitivity of a receiver will be discussed:

- frequency accuracy
- amplitude (level) accuracy
- error vector magnitude (for digitally modulated signals)

Frequency modulation deviation accuracy and frequency modulation distortion are two other specifications that affect sensitivity measurements.

Poor frequency accuracy will cause the signal to fall nearer to the skirts of the filters in the receiver; this will degrade the level of the signal and will reduce either the SINAD or BER measurement. A receiver will appear to have a lower sensitivity when tested with a signal that has poor frequency accuracy.

Slide #62

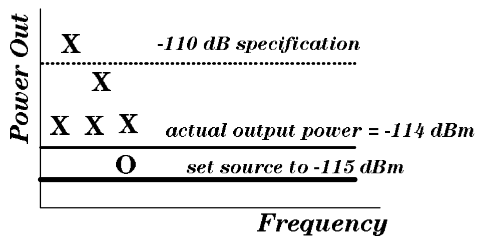
Applications and Critical Specifications

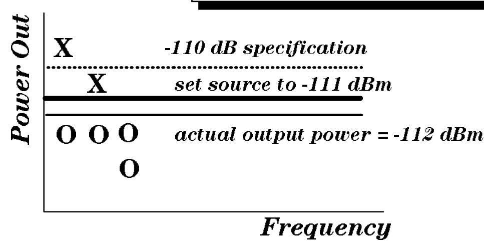
Receiver Sensitivity

- Level Accuracy

Customer is testing a -110 dB sensitivity pager:

X = Failed Unit
O = Passed Unit





Case 1: Source has ± 5 dB of output power accuracy at -100 to -120 dBm output power

Case 2: Source has ± 1 dB of output power accuracy at -100 to -120 dBm output power

When making a sensitivity measurement, the level accuracy of the signal generator is extremely important. For example, a pager receiver has a specified sensitivity level of -110 dBm. The measurement system will introduce error. The amplitude level accuracy of the signal generator supplying the test signal is the main source of error. In order to ensure that no receivers are passed with sensitivities that do not meet -110 dBm, the amplitude of the signal generator must be set below the receiver specification by an amount equal to the level accuracy of the signal generator. In this example, for a signal generator with a level accuracy of +/-5dB, the maximum amplitude of the test signal is set to -115 dBm.

In this example, the maximum signal generator level of -115 dBm ensures that only good receivers pass. However, not all receivers that are good will pass. For example, with a level accuracy of +/-5dB at a nominal level of -115 dBm, the output of the signal generator may be as high as -110 dBm (the receiver spec) or as low as -120 dBm. A good receiver may not work at -120 dBm but, because of the signal generator uncertainty, may be tested at -120 dBm. Of the six pagers tested in this example, only one passed.

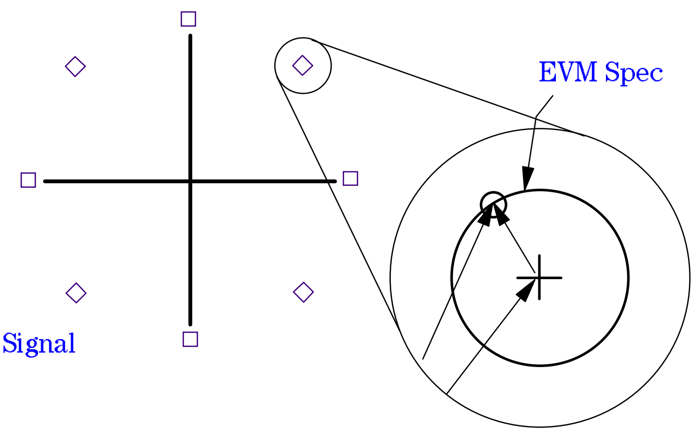
When tested with a signal generator that has a level accuracy of +/-1dB, the number of pagers that passed increases dramatically.

Slide #63

Applications and Critical Specifications

Receiver Sensitivity

- Error Vector Magnitude

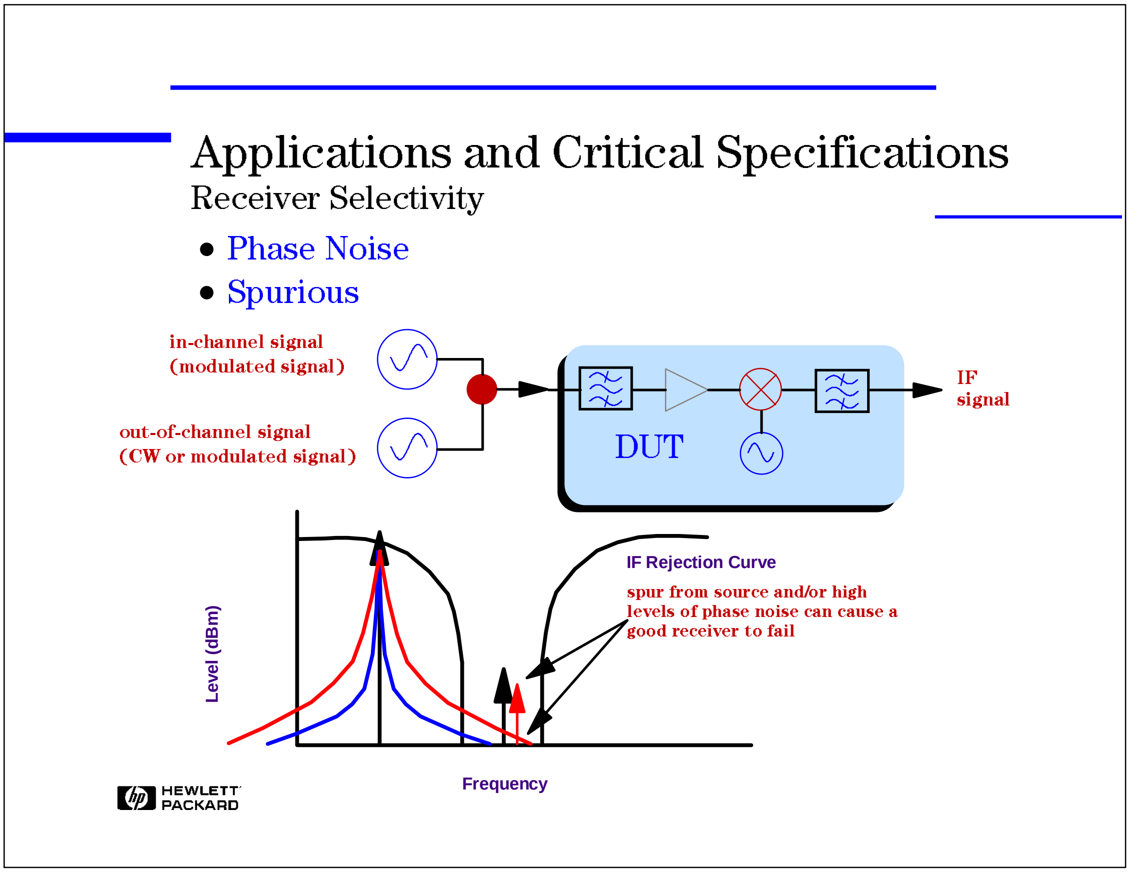


e.g. TETRA Signal
p/4 DQPSK
EVM < 1.0%

The best indicators of modulation quality are obtained from the constellation diagram of a signal. For phase shift keyed signals, the EVM measures the signal quality. For frequency shift keyed signals, the signal phase error is a more appropriate measurement. For amplitude shift keyed signals, magnitude error should be used.

For the above TETRA signal, the EVM is less than one percent. The size of the dots indicates the quality of the signal: A signal with a larger EVM would have larger dots.

Slide #64



Adjacent and alternate channel selectivity measures the receiver's ability to process a desired signal while rejecting a strong signal in an adjacent channel or alternate channel. This test is very important for communication receivers where channel spacings are narrow and many signals may be encountered in a small geographical area. An adjacent or alternate channel selectivity test setup is shown above. One signal generator inputs a test signal at the desired channel frequency at a level above the sensitivity of the receiver. The second signal generator outputs either the adjacent channel signal, offset by one channel spacing, or the alternate channel signal, offset by two channel spacings. The output of the out-of-channel signal is increased until the sensitivity is degraded to a specified level.

Frequency and amplitude (level) accuracy and the spectral characteristics of the test and interfering signal are important.

Poor frequency accuracy will cause the signals to be either too close or too far from each other and from the filter skirts. This can have the affect of appearing to improve or degrade the receiver performance.

We saw how level accuracy can affect the sensitivity measurement of a receiver. With two signals, the problems associated with inaccurate signals are compounded.

For FM receivers, the SSB phase noise of the interfering signal is the most critical spectral characteristic. The test is a measure of the performance of the receiver's IF filters. As the signal in the adjacent-channel is increased, the rejection of the IF filter outside the passband is eventually exceeded. If the phase noise energy inside the passband is detected, the receiver may appear to fail the test.

High levels of spurs can also degrade the selectivity measurement of a receiver. Signal generator spurs that fall within the passband of the receiver will contribute to the overall noise level in the passband.

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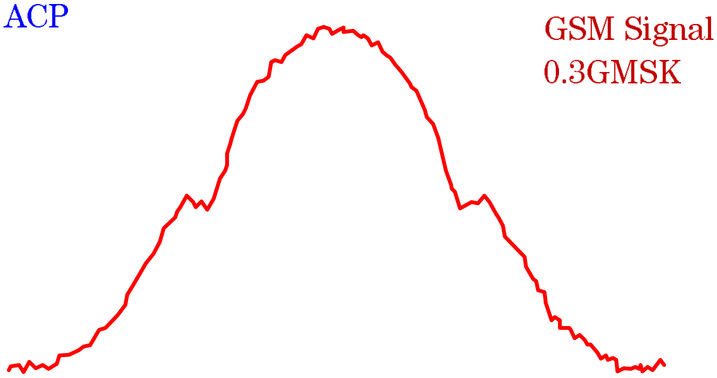
Slide #65

Applications and Critical Specifications

Receiver Selectivity

Spectral Accuracy:

- EVM
- ACP



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When measuring selectivity on receivers designed for digital signals, the most important spectral characteristic is spectral accuracy. With digital signals, the modulation sidebands are often wider than the channel spacings. These sidebands have considerably higher power at adjacent and alternate channel offsets than analog FM signals. Because of this, the phase noise and the spurious of the signal generator is much less important.

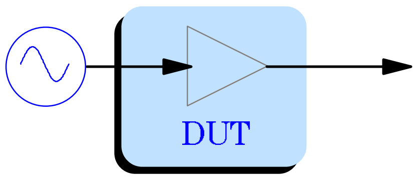
However, the spectral shape is very important. For the adjacent-channel interferer, the spectral shape should be exact. EVM is a good indicator of the spectral shape within the modulation bandwidth of a signal. However, even with a good EVM, a signal may have a significant amount of spectral splatter outside of the modulation bandwidth. Comparing the measured adjacent-channel power (ACP) to the theoretical ACP provides an indication of the spectral quality outside of the modulation bandwidth.

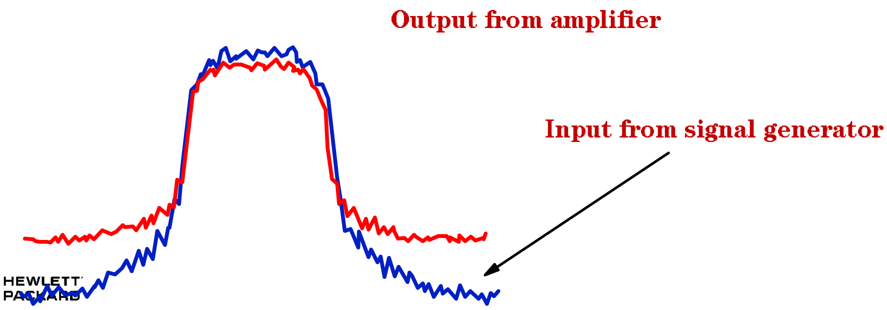
Slide #66

Applications and Critical Specifications

Spectral Regrowth

- ACP Performance





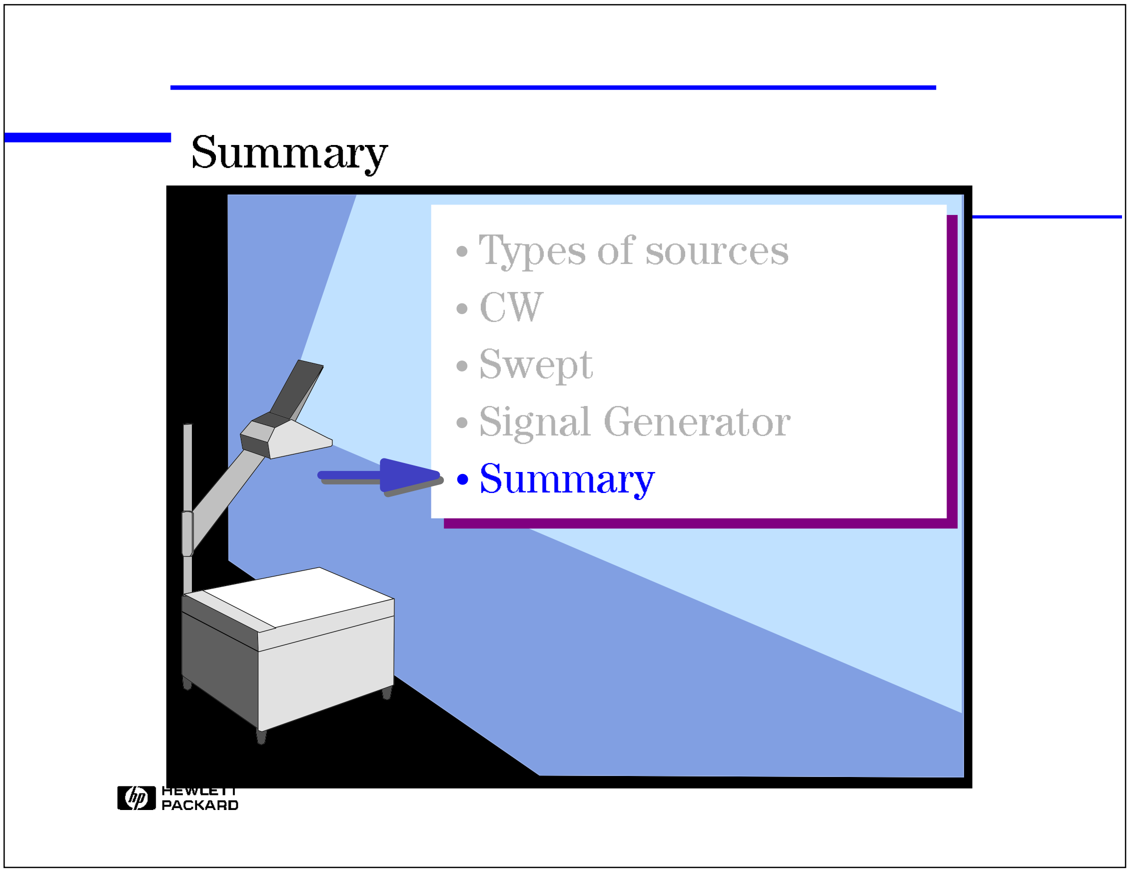
Spectral regrowth is a common amplifier test. When stimulated with a digitally modulated signal, the nonlinearities in amplifiers create shoulders in the output spectrum. These shoulders are referred to as spectral regrowth. Spectral regrowth is related to intermodulation distortion. Intermodulation distortion results from the interaction between two (or more) input sine waves. A digitally modulated signal can be expressed as an infinite sum of sine waves with properly weighted coefficients:

$$V(t) = \sum_{n=1} a_n \sin(2\pi f_n t)$$

The interaction of these sine waves produces the spectral regrowth.

Signal generators have amplifiers that can also introduce spectral regrowth. When measuring an amplifier, the spectral regrowth, or ACP performance, of the signal generator should be less than that of the amplifier under test.

Slide #67



This seminar reviewed the basics of sources. Aspects of the design, specification, and application of sources were explored. A brief list of some of the reference material has been inlcuded for further study. In addition, Hewlett-Packard has an extensive library of application notes that provide more information on how to use sources.

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