Agilent Technologies

Overcoming Mixer Measurement Challenges

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presented by:

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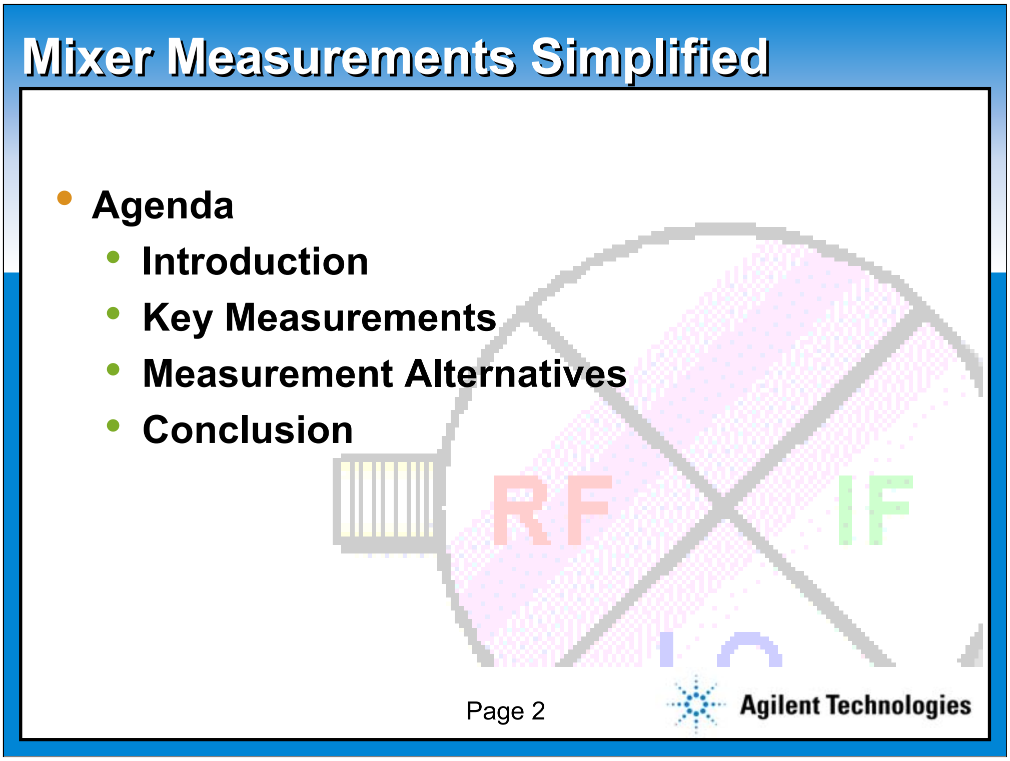
Today we will be looking at overcoming measurements challenges associated with frequency translating devices (FTD) such as mixers and converters. Frequency translating devices are at the core of all of today's RF and microwave wireless communication systems. These devices present unique measurement challenges since input and output frequencies differ, requiring different measurement techniques than those used for linear devices such as filters and amplifiers.

The presentation today is broken into two parts.

If you are new to mixers design and test. If you are looking for the measurements suite to characterize a mixer. Or if you are just looking for a good review of mixer measurements section one is for you. Mixer measurements simplified looks at the key measurements that need to be made on mixers and what they mean. We will examine several alternative methods of testing mixers and discuss their pros and cons for different types of measurements.

If this all sounds familiar because you have been testing mixers for years, hang on, section two is for you. vector corrected frequency-offset measurements dives more deeply and technically into using a vector network analyzer in frequency-offset mode of operation. We will reveal two new calibration processes that Agilent introduced at European Microwave 2002 in Milan, Italy. Together these correction techniques offer a new level of accuracy for magnitude and phase measurements. These techniques, implemented in the PNA frequency-converter application, simplify complex and error prone mixer measurements.

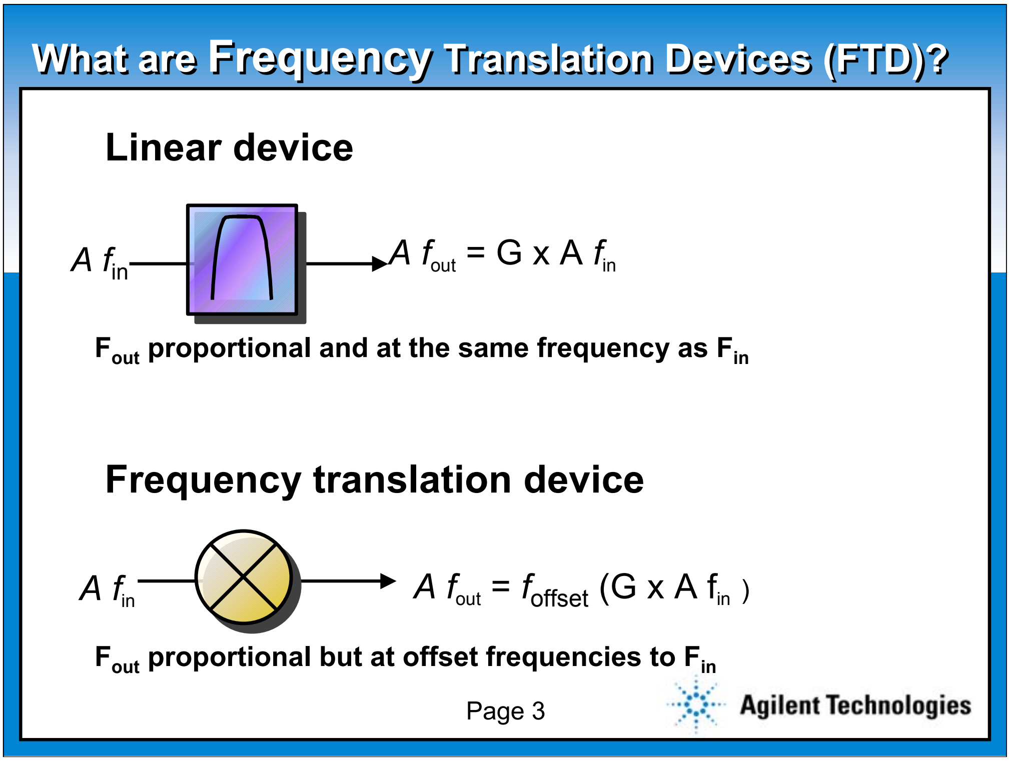
We are excited to share this with you today. So let’s get started.



We will start by defining exactly what we mean by a frequency translating device. It is important to briefly present an overview of their expected mode of operation. This way, we will know that to expect when setting up of and performing measurements.

Next, we will look at the key measurements that are needed to characterize these devices. We will start with transmission measurements including both magnitude and phase. Then, we will look at the reflection and isolation measurements. This section will conclude with a brief look at some nonlinear measurements that are also important.

Finally, we will conclude the presentation with a look at a variety of different measurement solutions. Some solutions are of value for a limited set of measurements, while others address a more complete complement.

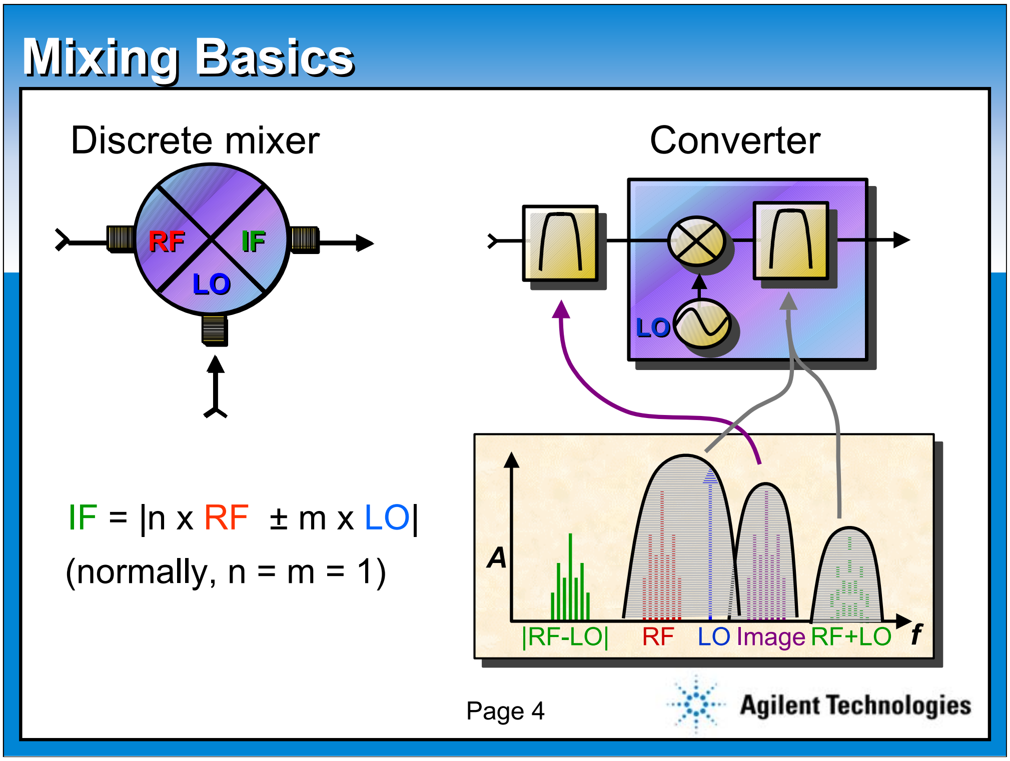


Frequency translating devices are non-linear devices by definition.

There are two definitions of non-linearity.

A linear device, such as a filter, will only affect the amplitude and phase of a transmission signal. Its purpose is to attenuate certain bands of frequencies while allowing others to pass with minimal loss. Conversely, non-linear devices also affect the frequency of an incident signal. This is the first definition of non-linearity. If a frequency appears at the output that was not present at the input it is said to have non-linear effects. An amplifier is also a device that has non-linear effects, but great aims are made to operate it within its linear region. When it is driven beyond its linear region it produce harmonics of the input fundamental. Instead of only producing harmonics, which are integer multiples of the input frequency, a FTD is designed to utilize its non-linear effects to shift the input signal by a fixed amount. The term "translation" refers to this frequency shift, and the overall process is called conversion. When the output signal is lower in frequency than the input signal, the FTD is said to down-convert the input signal. Conversely, a higher output frequency is a result of up-conversion.

Another characteristic of a linear device is that output response is directly proportional to the input. Except for translation, FTD are expected to behave linearly. Just as filters and amplifiers, they exhibit gain or loss, a particular frequency response, reflection characteristics, and all of the other performance parameters of linear devices. They also exhibit many of the same undesired non-linear behaviors of amplifiers such as compression, and intermodulation distortion.

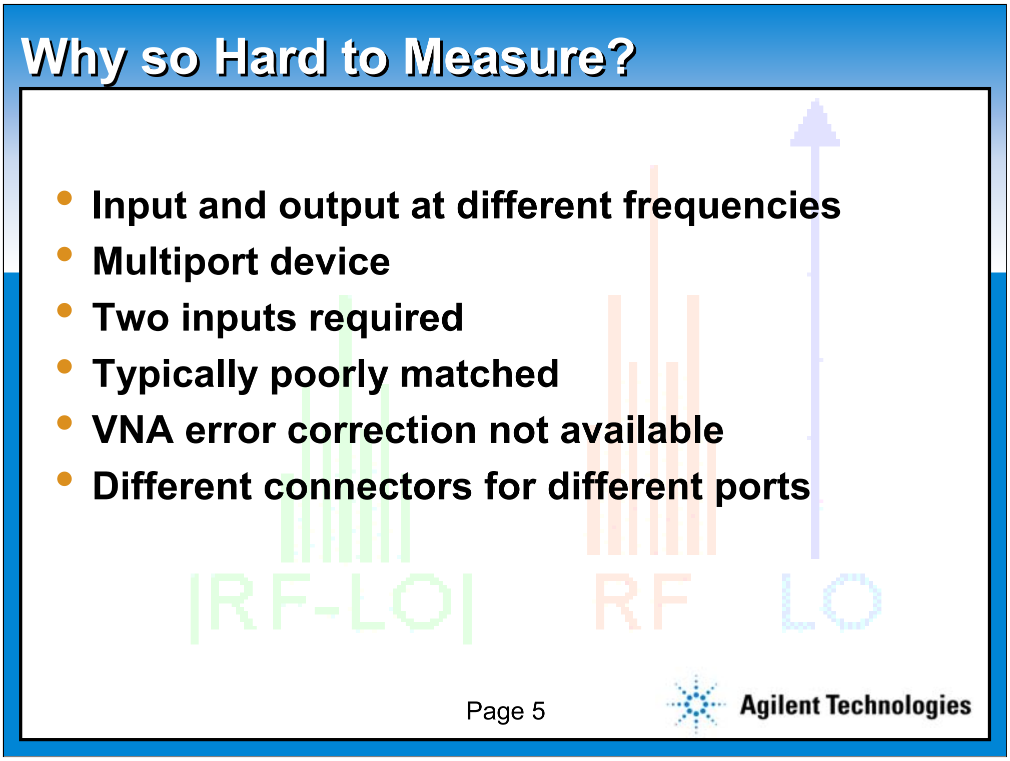


A mixer is the most general class of frequency translation devices. It is a three port device. Its input and output ports are commonly referenced as RF (radio frequency) and IF (intermediate frequency). The third port exists for the purpose of controlling the frequency translation effects. It is referenced as the LO (local oscillator) port because another frequency source is supplied there. Depending on whether the mixer is being used for up or down conversion will determine which port is the input. For the purposes of this prsentation we will be using a down-converter for the illustration so the input port will be the RF and the output down-converted to an IF. The non-linear mixing process produces the sum and difference frequencies of the LO and RF signals (RF ± LO). The output or IF of a mixer consists primarily of these two signals. Usually only one is desired, and the other is filtered away.

Many other smaller, higher-order signals appear at the IF port as well. These are due to harmonics of the LO and RF signals mixing within the mixer and their sum and difference signals produced as well. These products are also typically filtered out. In general, the mixer will produce all signals of the form (n*LO ± m*RF), where n and m are integers.

The word converter is most often used interchangeably with mixers to describe any device that produces a frequency translation. In general converters integrate other component in one module.

When a module has an internal LO, there is no need for the third port. A complex frequency translating device such as a receiver might have multiple conversion stages and more then one internal LO, but from the outside it behaves as a two-port device having some composite frequency shift. Most two-port devices have internal LO that are inaccessible, and in some cases, they require different test techniques than those used for three-port devices such as mixers. In particular, phase measurements are more difficult since the LO is unavailable to provide a synchronous reference..



A vector network analyzer is the standard choice for measuring components. So it might seem like the obvious choice for mixers as well. But there are several characteristics that make mixer test more challenging. As we've seen they are inherently non-linear devices where the desired input and output frequencies are different.

They are multiport devices which require 2 input signals at different frequencies with one or both sweeping in frequency.

They are typically poorly matched to 50 Ohms. This will cause errors signals to enter into the measurement from reflections off the mixer and the test system. Since mixers are non-linear, the well defined 2-port vector error correction found on vector network analyzers is not valid and can not be used to correct error signals.

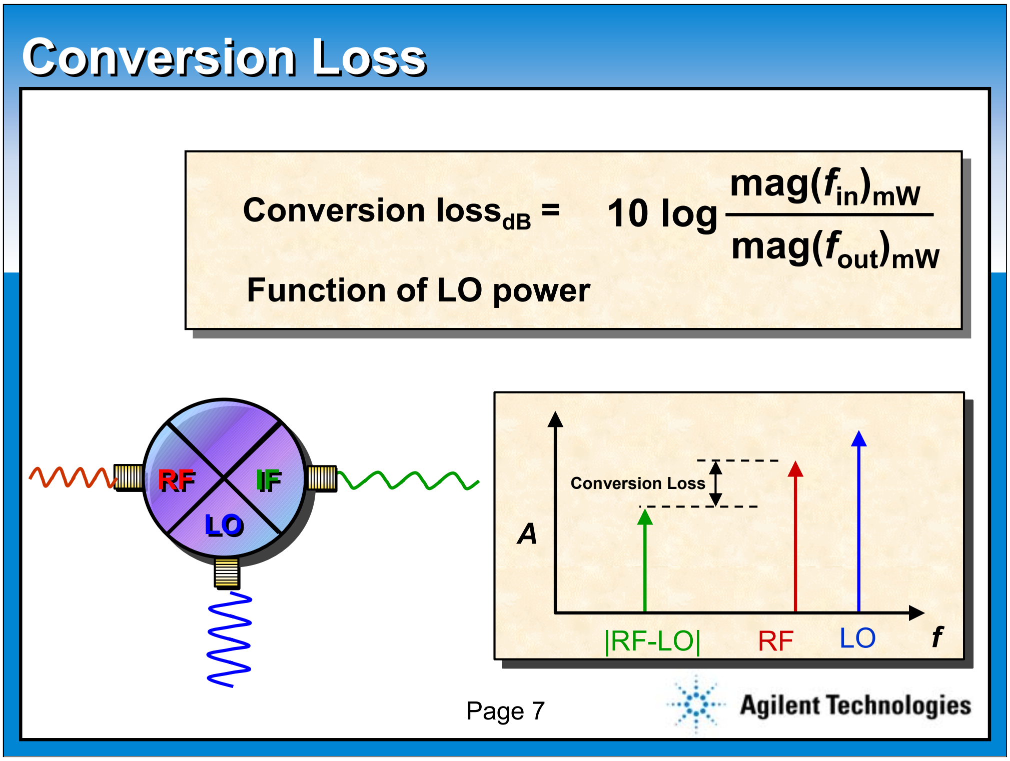
Since one of the three ports is in a different frequency range than the others, the mixer can typically have a different connector type, making them non-insertable.

As we will see in the measurement section of this presentation both the linear and non-linear response of the device needs to be characterized.

Because of all of these challenges, network analyzers have become only one of many alternative methods of testing frequency translation devices. There is no one size fits all solution. Each solution tries to address some of the challenges listed here and some prove to be more beneficial for certain measurement and not valid for others. We will explore several of the alternative test configurations that are currently being used. But first, let's explore the measurements themselves that help us understand both the desired and undesired behaviors of a mixer.

Key Measurements

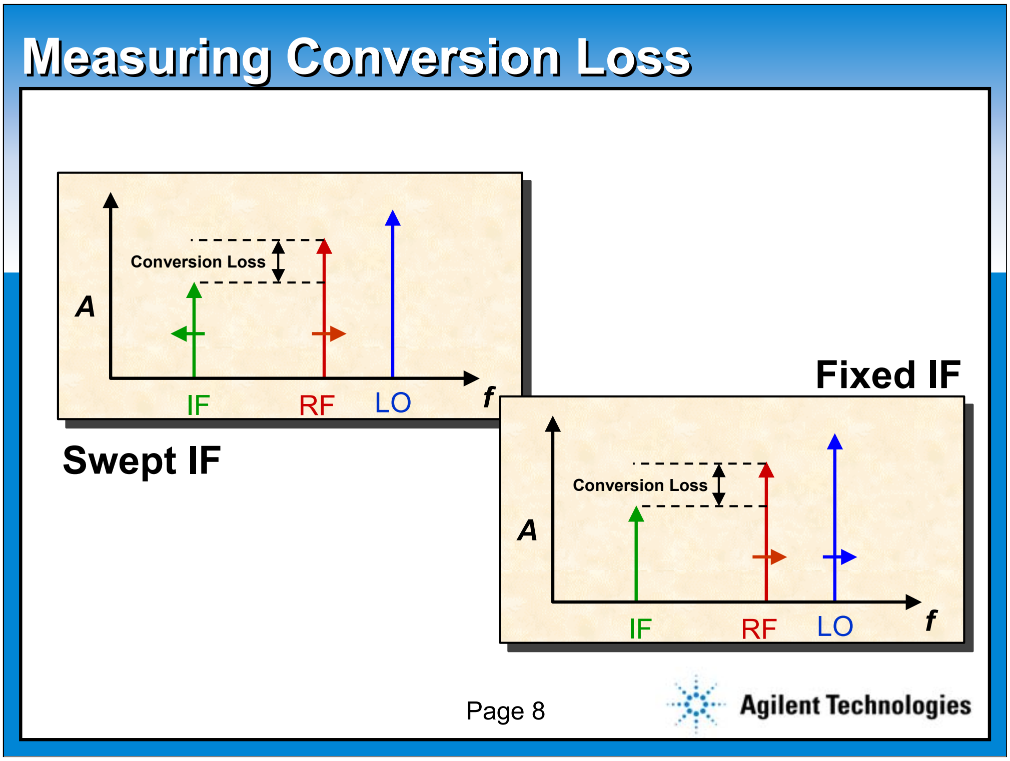
- Transmission
  - Conversion Loss
  - Phase
  - Group Delay
- Reflection
  - VSWR
  - Return loss
- Isolation
- Non-Linear
  - Conversion compression
  - Higher order mixing products
  - Two- tone IMD



The first measurement we will cover is conversion loss. In the case of a modules with integrated amplifiers we will see a conversion gain. Ideally, it is desirable to convert all the power applied at the input frequency to power at the desired output frequency. However, mixers by their non-linear nature divide the output power between the sum and difference frequencies, as well as higher-order mixing products. Conversion loss is a measure of how efficiently a mixer converts energy from one frequency to another, and is one of the most important specifications on a mixer.

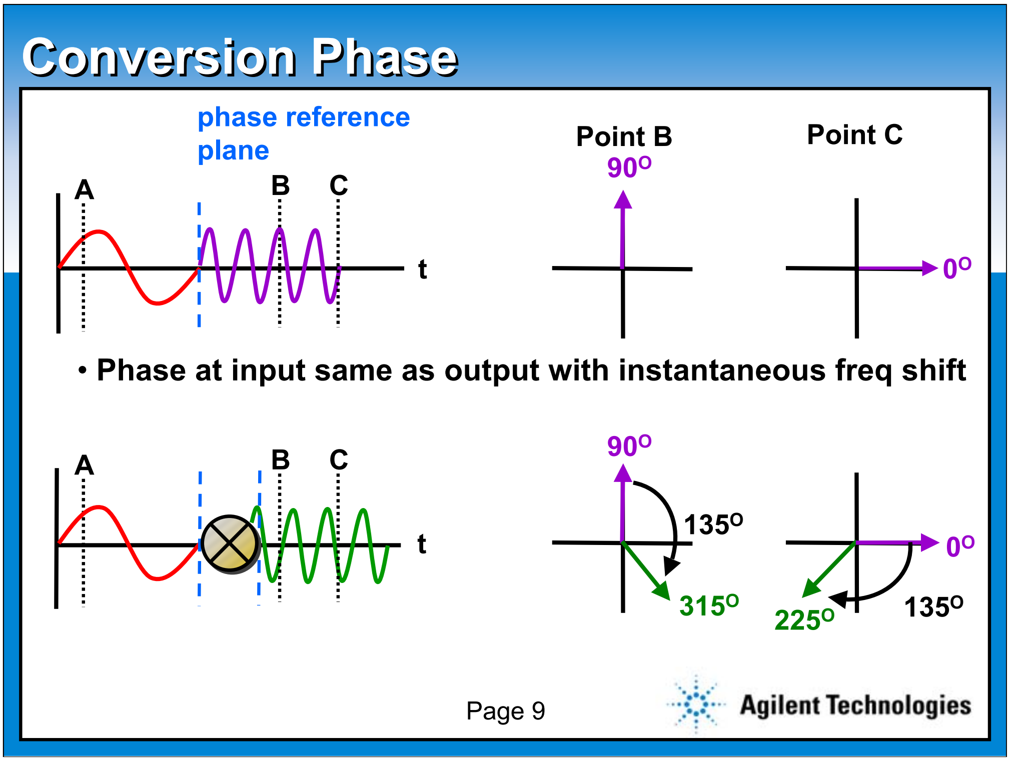
Conversion loss is defined as the ratio of CW input power to the CW output response power expressed in dB, for a given amount of applied local oscillator power, and it is usually measured versus frequency. While conversion loss of a mixer is usually very flat within the frequency span of its intended operation, the average loss will vary with the level of the LO.

Since converters integrate amplifiers and filters, an important parameter is their bandwidth and gain flatness within a specified bandwidth; usually set by one or more internal filters.



The easiest type of conversion loss or gain flatness measurement is to sweep the RF input and keep the LO frequency constant. This is called a swept IF measurement. This measures the conversion loss over the full operating range of the mixer and not just at the center of it bandwidth. It can be said to operate with a certain performance over the whole band. Notice when the LO is above the RF, known as high side mixing, the IF will sweep in the oppose direction from the RF.

If the FTD has an integrated IF filter, it might be tested by holding the IF fixed and having the LO track the RF signal. In order to accomplish this, the LO must sweep in conjunction with the RF input signal. In many cases, this measurement more closely matches the operation of the DUT in the actual application.

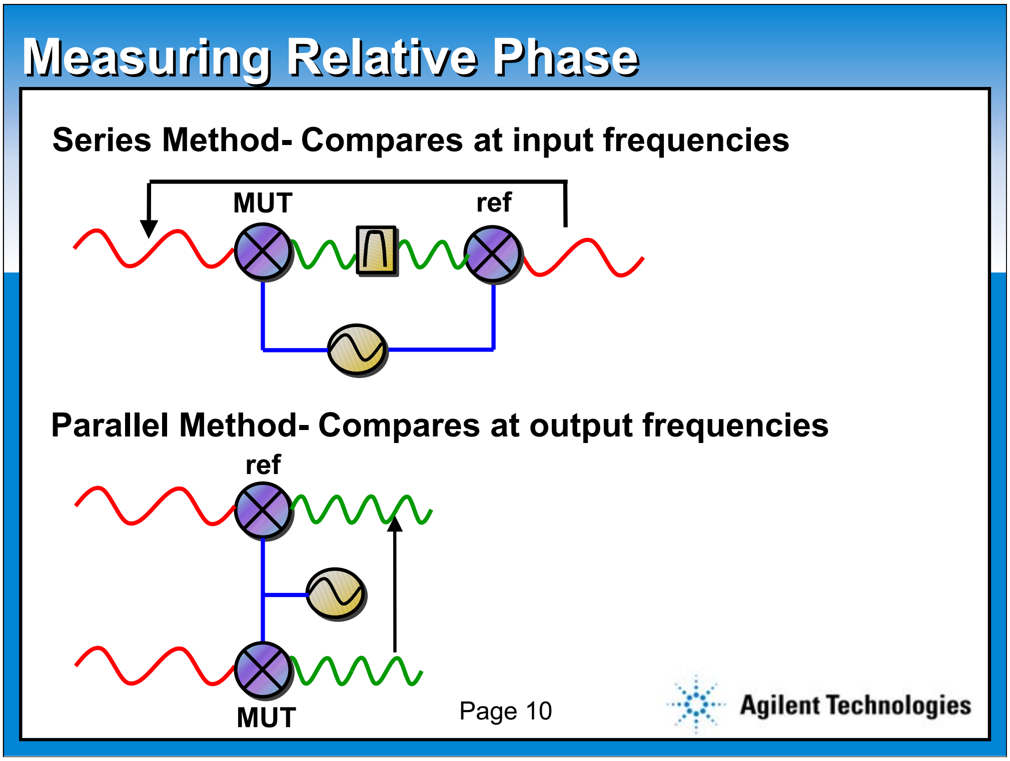


It is also of interest what the phase is doing as it passes through the mixer.

Since the input and output frequencies are not the same, the definition of conversion phase can be confusing. We define the conversion phase as the phase shift of the output compared to the input when the input synchronously converts to the output frequency with an ideal, zero phase shift.

Remember, the definition of conversion loss was the input power level compared to the output power level, but was a function of the LO as well. Likewise, the phase of the output signal is a function of the input phase and the LO phase. Therefore, the reference signal must be converted in frequency with a constant phase relationship to the signal it will be compared to. This is referred to as synchronous conversion. Any different frequency shift between the two signals will cause a phase measurement error. This is the primary challenge when measuring converters with internal LO that their phase can not be referenced to.

In this example the input (red) signal can not be compared directly to the output (green) signal. Only after the output reference (purple) is created can the desired output signal phase be measured. Once the reference is created it can be used as a reference regardless of where the signal is compared. Its phase shift or difference will remain the same. Notice that at sample point B and C both return the same phase shift of 135 degrees independent of the point chosen to make the comparison.



There are a number of techniques that have been developed to approximate the theoretical synchronous zero length frequency shifter. Most techniques seek to compare the known phase shift of a “golden” mixer to the phase shift of the mixer under test. This gives a phase shift relative to the reference. The extent to which the “golden” mixer is accurately characterized adds to the overall accuracy of the measurements.

This practical example shows two methods of accomplishing a relative phase measurement.

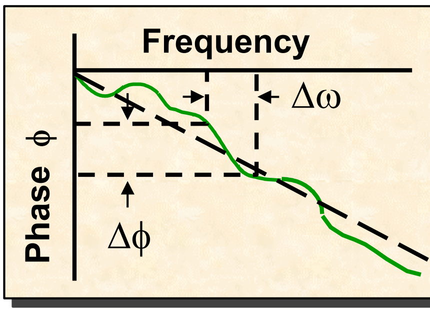
In the top example a “golden” or reference (ref) mixer is placed in the measurement path with the mixer under test (MUT). The reference mixer has a well known phase and amplitude response over the frequency of interest. It is used to reconvert (down convert) the output of the MUT to the input frequency where its shift can be compared to the signal incident on the MUT. A filter is required to remove the unwanted image, so they will not distort the response of the desired signal. The filter response must be removed from the pair’s response, and the mismatch between the filter and the mixers can cause additional uncertainty. Any phase delay of the reference is subtracted out, the best that it is known, and the remaining variation is attributed to the MUT.

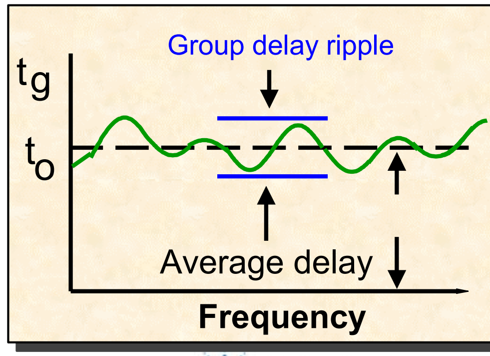
The bottom method uses a parallel reference (ref) mixer approach. It also uses the same LO for both the reference and MUT mixers to achieve the synchronous requirement. In contrast, the reference signal is converted at the same time as that of the mixer under test and the output phases are compared. Again, the phase delay of the reference mixer must be known and accounted for.

Group Delay

- **Ripple indicates phase distortion**
- **Average delay indicates electrical length**
- **Aperture setting is important**

$$\textbf{Group delay}_{(t)} = \frac{-d\,\phi}{d\,\omega} = \frac{-1}{360} \times \frac{d\,\phi}{d\,f}$$





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Of more interest than the phase shift through the device is how linear with frequency the phase shift remains. Group delay is such a measure. It is especially important in digital communications because slight changes in the phase relationship between the frequency components of a pulse can drastically change its shape. Satellite transponder manufacturers are especially concerned with group delay because their devices must receive and transmit a wide variety of digital signals over a very broad frequency range. Mixers often have baluns that affect their phase response near the limits of their operating frequency ranges.

Group delay is difficult to measure on frequency translating devices because the input and output are at different frequencies--there is no reference against which the output phase change can be compared.

Group delay is calculated by differentiating the insertion-phase response of the device under test (DUT) versus frequency. Another way to say this is that group delay is a measure of the slope of the transmission phase response. The linear portion of the phase response is converted to a constant value (representing the average signal-transit time) and deviations from linear phase are transformed into deviations from constant group delay. The variations in group delay cause signal distortion, just as deviations from linear phase cause distortion. Group delay is just another way to look at linear phase distortion.

When specifying or measuring group delay, it is important to quantify the aperture in which the measurement is made. The aperture is defined as the frequency delta used in the differentiation process (the denominator in the group-delay formula). As we widen the aperture, trace noise is reduced but less group-delay resolution is available (we are essentially averaging the phase response over a wider window). As we make the aperture more narrow, trace noise increases but we have more measurement resolution.

* Generally, group delay measurements are not normally performed on mixers and converters used in pt-to-pt and pt-to-multi-point radio (e.g., LMDS). Given the slow fading characteristics of line-of-sight signals, the adaptive equalization in the receiver has enough horsepower to compensate for both fading and internal systematic errors.

Key Measurements

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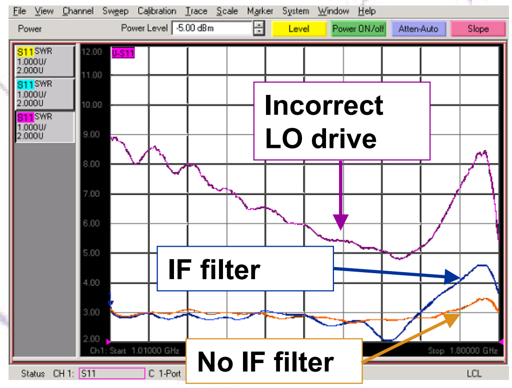
- Isolation
- Non-Linear
  - Conversion compression
  - Higher order mixing products
  - Two- tone IMD

Reflection- VSWR and Return Loss

- Reflection depends on conditions at other ports
- Must use correct LO power
- Must terminate unmeasured ports with correct impedance



Now that we have covered measurements for transmission, let's move on to reflection. Reflection measurements are linear, even when testing frequency translating devices, since the reflected signal does not undergo a frequency shift. Therefore, these measurements are essentially standard S-parameter measurements, the same as for filters and amplifiers, with a few minor variations.

Mixers are three-port devices, and the reflection from any one port depends on the conditions of the other two ports. When measuring reflection on a three-port device, it is important to terminate the ports not being tested with the impedance that they would see during actual operation. While this is often the characteristic impedance, Z_o, it could also be something more complex. For example, if the IF port of the mixer will be directly connected to a filter, then this filter should be used when testing RF or LO-port reflection.

When testing the RF or IF ports, the LO signal should be applied at the power level that will be used in actual operation. Since the bias point and impedance of the mixer diodes is a function of LO level, the reflection at the other ports will also be a function of LO level.

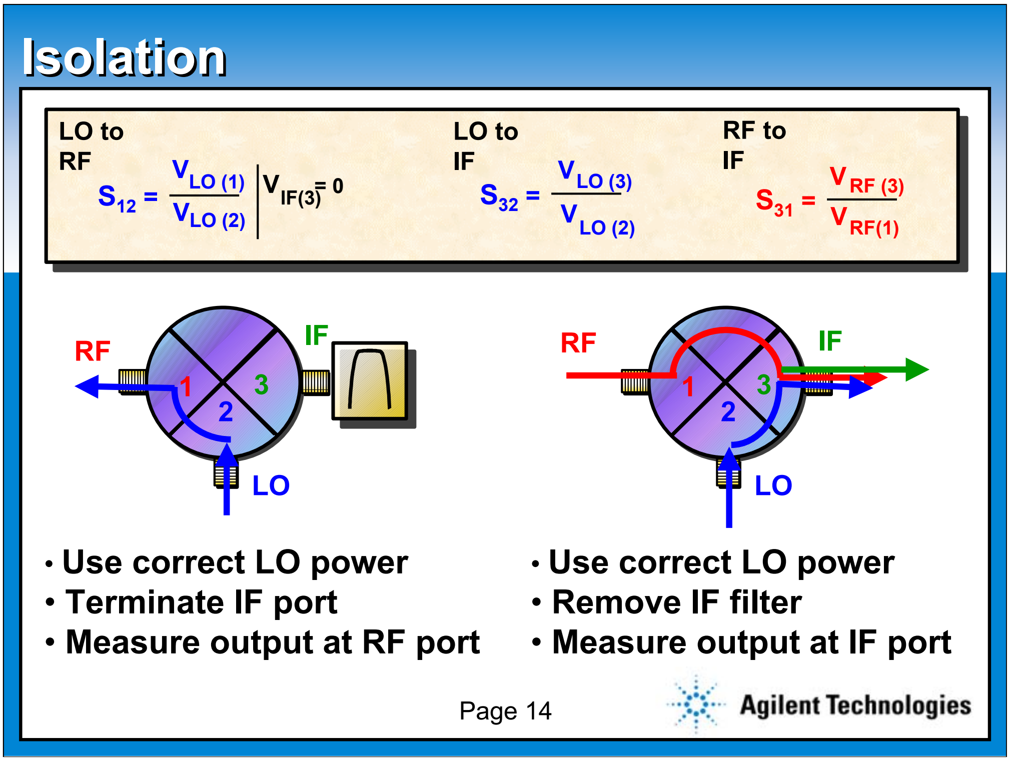
This graph shows three SWR measurements on the input port of a RF mixer.

All measurements used a 1-port calibration.

The orange trace was measured with a LO drive of +10 dBm and a input power of –5dBm without an IF filter at the output port.

The blue trace uses the same input and LO drive but places a 2.5 GHz lowpass IF filter on the output for the mixer.

The purple trace keeps the IF filter but decreases the LO drive to 0dBm. It can be seen how much this 10 dBm decrease in LO drive effects the SWR measurement at the input of mixer.



Isolation is a measure of the leakage, or feed through, from one port to another. The more isolation a mixer provides, the lower the amount of feed through.

Isolation is a transmission measurement with the stimulus and the response measured at the same frequency. In the example mixers on this slide, the ports are labeled RF port 1, LO port 2, and IF port 3. The three isolation terms of interest are S32 LO-to-IF isolation, S12 LO-to-RF isolation, and S31 RF-to-IF feed through.

Measuring RF-to-IF feed through of a converter is identical to that of a mixer. The RF-to-IF feed through is generally very small for a converter due to the inclusion of an IF filter in the device. Because of this, the measurement may require increased dynamic range and a lowered noise floor.

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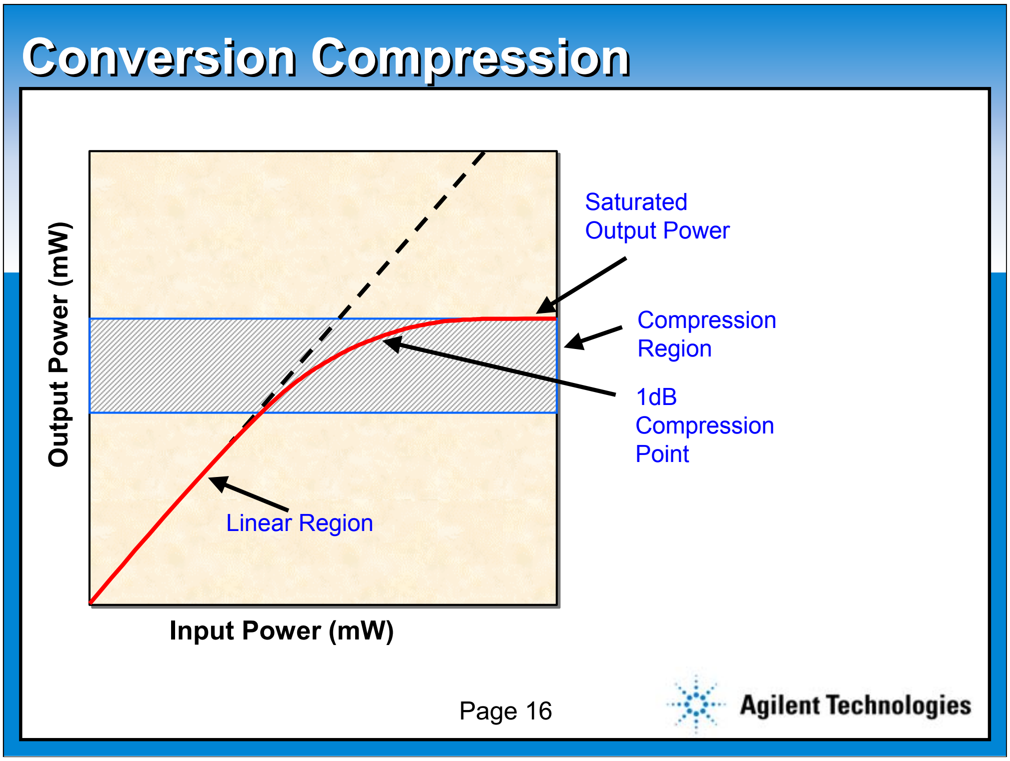
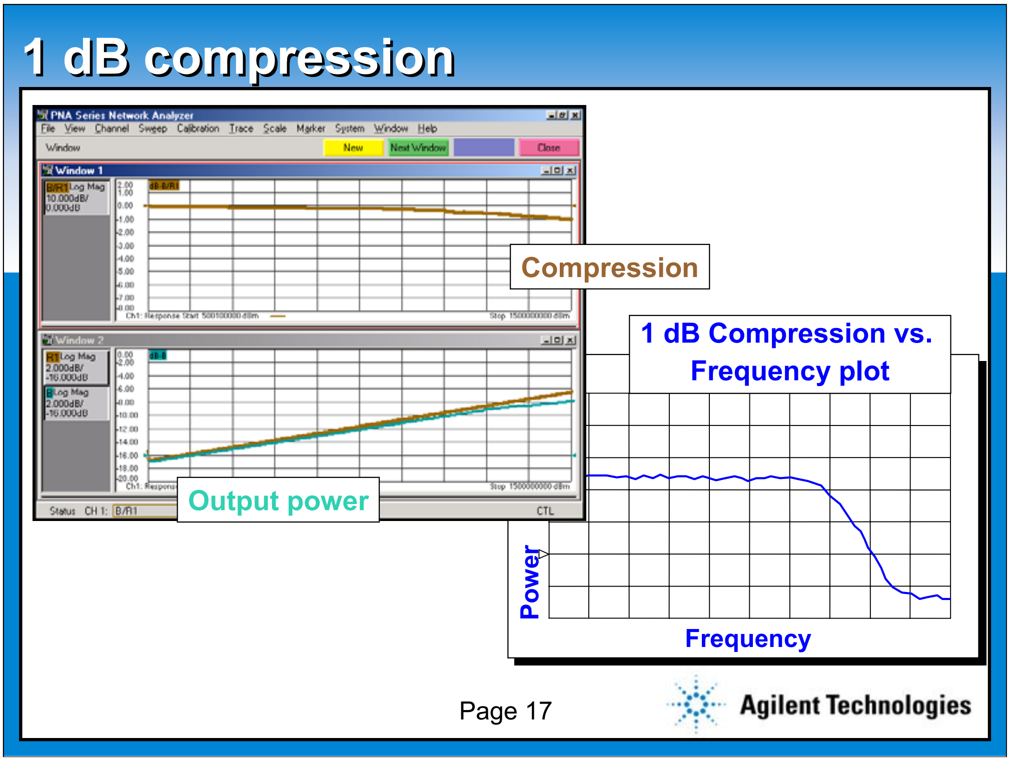


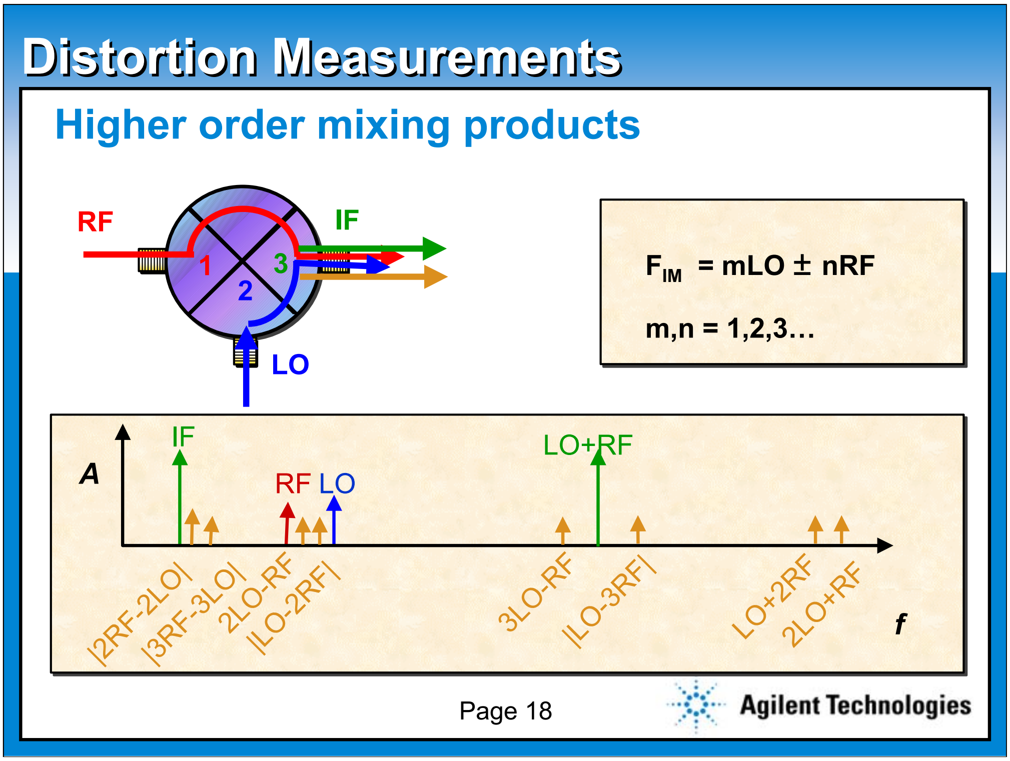
It is important to understand some of the undesired non-linear effects that can be introduced when a mixer is driven beyond its linear region.

FTD have a compression region where a change in the level of the RF input results in a smaller change at the output. The point where compression starts to occur is directly related to the LO power level and the physical configuration of the mixer. Since converter modules typically have both mixers and amplifiers, there is no easy way to predict where compression will occur. Shown above is a plot of output power versus input power at a single frequency. The linear region of operation is where the gain is constant and is independent of power level. The gain in this region is commonly referred to as “small signal gain.” At some point as the input power is increased, the gain appears to decrease, and the amplification is said to be in compression. Under this non-linear condition, the output is no longer sinusoidal, resulting in some output power being presented in harmonics rather than occurring only in the fundamental frequency. As the input power continues to increase, the amplification becomes saturated, and the output power remains constant. At this point the gain is essentially zero, since any further increase in input power results in no change in output power.



The most common measurement of compression is the 1 dB compression point. It is defined here as the input power level which results in a 1 dB increase in loss, or a 1 dB decrease in gain for converters with amplification. The easiest way to determine the 1 dB point is to sweep power at a certain CW frequency and measure the conversion loss. The top measurement shows both the input and output power measurements being swept from –10 dBm to 2 dBm. The top window in that measurement is the conversion loss normalized to a linear portion of loss and then divided by the conversion loss as power is swept. The conversion loss increase by 1 dB at about +.25 dB input power. The flat part of the trace in the upper left is the linear, small signal region, and the curved part on the right side corresponds to compression caused by higher input power.

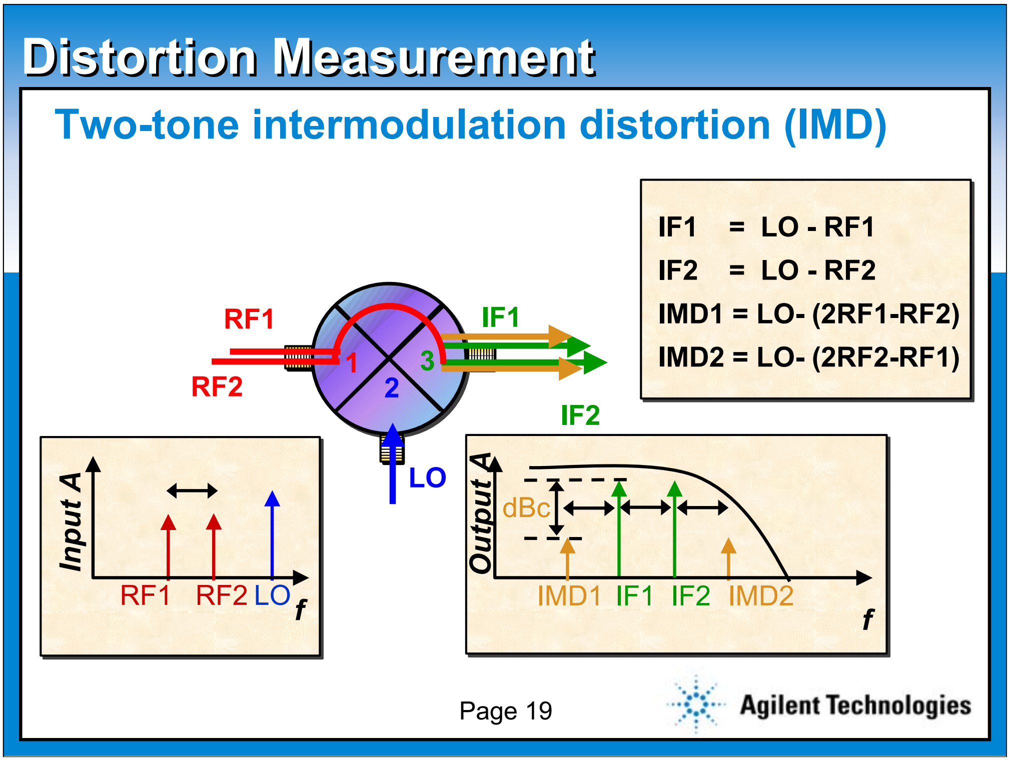
This measurement is made at a single frequency. It is also important to determine at which frequency the compression first occurs and how that compression changes with frequency. Manually or through automation a series of 1 dB compression measurements can be made. Their results can be plotted to get a compete picture of the 1 dB compression point power at each frequency of interest.



We have looked at the effects of compression as a mixer goes beyond its normal operating region. There are two other undesired nonlinear effects we need to take a look at.

The first measurement is a look at the additional mixing products that are produced when the harmonics of the RF and LO mix together. This effect is referred to as higher order mixing products. From the output spectrum above we can see a rather large number of additional mixing products. In reality there are many more then this. Any harmonic of the RF can mix with any harmonic of the LO to produce a spurious signal. M and n do not have to be equal. When they are equal a special case arises. The spurious signal produced will be the same harmonic away from the IF as the one that mixed. For example IF = (LO-RF) then 2LO-2RF = 2(LO-RF) = 2IF and 3(LO-RF) = 3IF. If they are not removed with filtering before the signal is amplified they will be added to a harmonics produced in the amplifier itself.

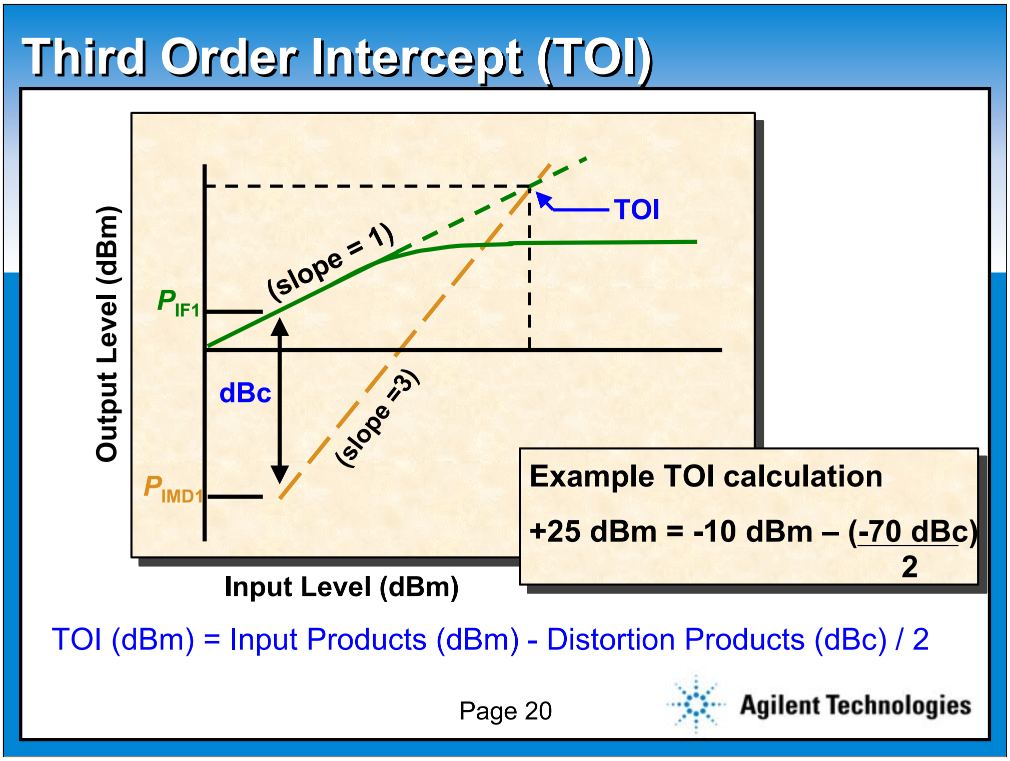
Measuring all of these spurs are one reason mixer test will always be difficult and somewhat custom. The good news is that they are very predictable. We know where we need to look to find them. In general, these signals are undesired and they are measured to insure that their level is appropriately low.



Another undesired nonlinear response is intermodulation distortion (IMD). Intermodulation distortion is a common problem in narrow-band systems. This is a measurement that is generally not preformed on discrete mixers, but on converters that integrate multiple discrete components. When two (or more) signals are present in a system, strong harmonic components are often generated. Both RF tones will fundamentally mix with the LO to produce an IF1 and IF2 at the output. As we saw in the previous slides, the harmonics of the RF and LO will also mix to produce higher order mixing products. These distortion products can degrade the performance of communication systems by causing distortion in analog systems and symbol errors in digital systems. Signals transmitted with excessive third order IMD can interfere with other transmissions. Receivers must also be distortion-free, especially in the preamplifier stages, to prevent crosstalk between adjacent channels. Most of the mixing products fall out of the IF band and can be easily filtered out. In this case the third order terms are of most interested because they will fall very close to the fundamental RF tones.

For example, two sinusoidal RF signals spaced 1 kHz apart will produce third-order distortion products that are 1 kHz above the higher IF signal and 1 kHz below the lower IF signal. These distortion components fall within the actual communication band of interest, and they typically cannot be removed by filtering.

Third-order distortion is either specified as the intermodulation products in dBc resulting from the input signals, or as a figure of merit called third-order intercept (TOI). A typical third-order intermodulation specification might specify that the third-order products be < -75 dBc for two -10 dBm RF signals at the input.



TOI is based on the relationship that on a logarithmic scale, the amplitude of the third-order products rise three times as fast as the amplitude of the fundamental signals. For example, if the fundamental signals increased 10 dB, the third-order products would increase 30 dB. In theory, as the amplitude of the fundamental signals increases higher and higher, there would exist a point where the amplitudes of the fundamental signals and the third order products would all be equal. This extrapolated point of intersection is called the third-order-intercept point.

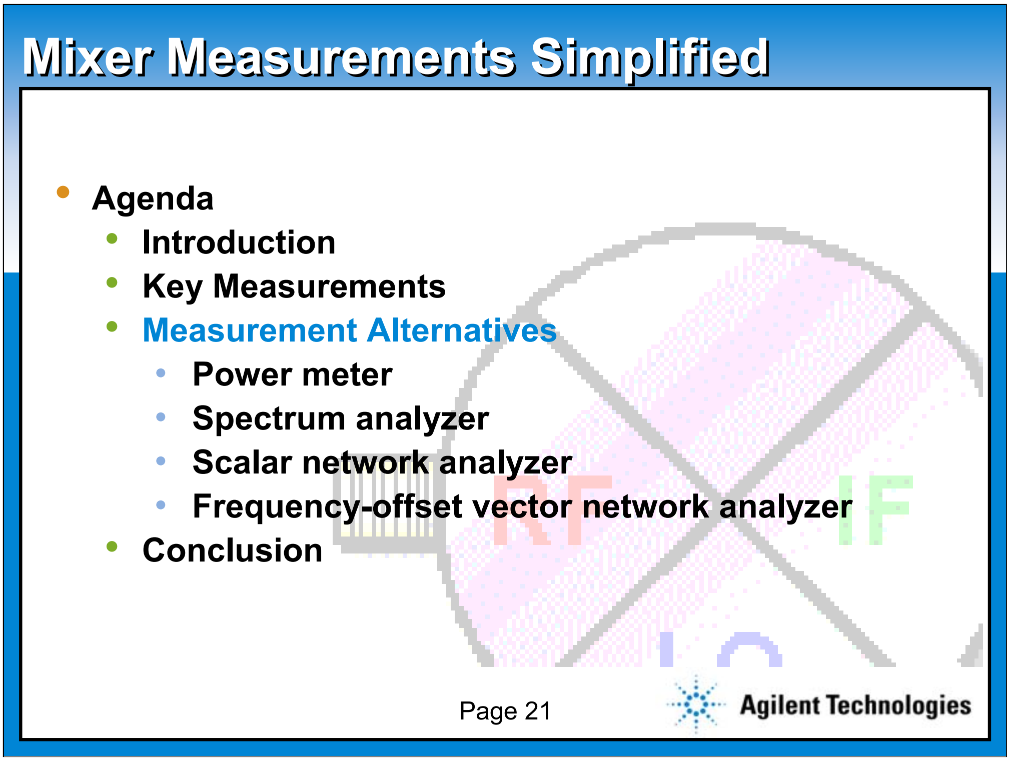
This condition can never be achieved in practice, since the analyzer would be in severe compression at that point. TOI can be calculated from lower-level amplitude measurements as follows:

TOI (dBm) = Input Signal levels (dBm) - Distortion products (dBc) / 2

For example, if two -10 dBm input signals produced third-order products that were -70 dBc, TOI would be -10 - (-70 / 2) = +25 dBm. TOI is sometimes referenced to the power of the two fundamental signals at the output, instead of the input.

TOI is a constant value independent of input power, and can be used to predict the level of distortion products at any input power level. If TOI is known, harmonic levels can be computed as follows:

Distortion Products (dBc) = 2 * [Input Signal level (dBm) - TOI (dBm)]



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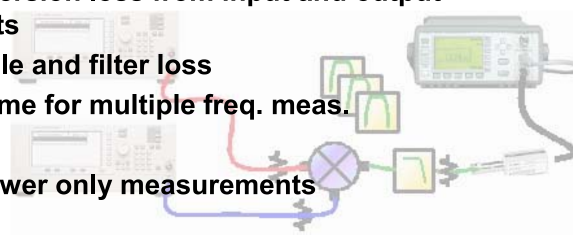
Power Meter with 2 sources

Measurement capabilities:
- Power

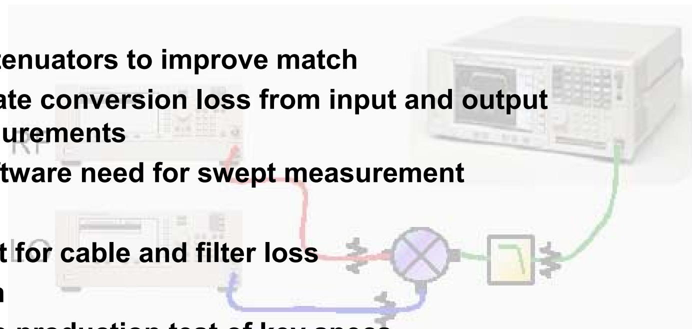
- Advantages
  - Very accurate power measurements
  - Inexpensive
- Limitations
  - Requires filters to isolate correct signals
  - Requires attenuators to improve match
  - Must calculate conversion loss from input and output power measurements
  - Must correct for cable and filter loss
  - Slow and cumbersome for multiple freq. meas.
- Best application
  - Single frequency power only measurements



Spectrum Analyzer with 2 sources

Measurement capabilities:
- Power
- Isolation
- Distortion

- Advantages
  - Easy setup
  - Able to view multiple signals simultaneously
  - Noise figure meas. often available
- Limitations
  - Requires attenuators to improve match
  - Must calculate conversion loss from input and output power measurements
  - External software need for swept measurement
  - Slow
  - Must correct for cable and filter loss
- Best application
  - High volume production test of key specs
  - Locating M and N spurs

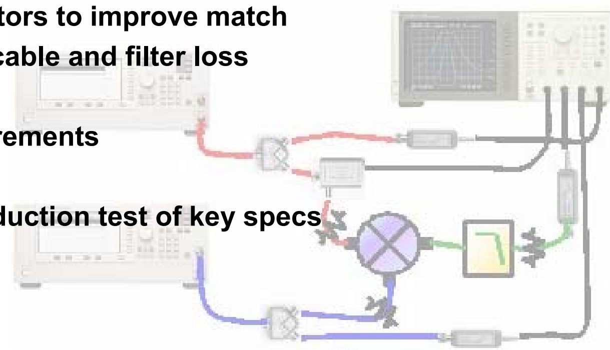


Scalar Network Analyzer with 2 sources

- Advantages
  - Makes ratioed measurements eliminates calculations
- Limitations
  - Complex setup
  - Requires attenuators to improve match
  - Must correct for cable and filter loss
  - No calibration
  - No phase measurements
- Best application
  - High volume production test of key specs

Measurement capabilities:

- Conversion loss
- VSWR
- Conversion Compression
- Isolation



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Traditional

Frequency-offset VNA with 1 source

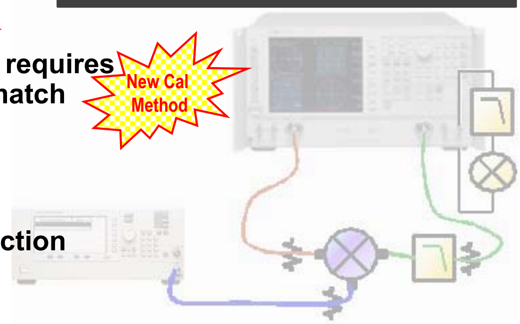
- Advantages
  - Able to make all key meas. magnitude and phase
  - Makes calculation directly
  - Calibrates out losses
- Limitations
  - Complex setup
  - Limited error correction, requires attenuators to improve match
- Best application
  - Design verification
  - When filter is integrated
  - High performance production testing of key specs

Simplified

New Cal Method

Measurement capabilities:

- Conversion loss
- VSWR
- Conversion Compression
- Isolation
- Distortion



Conclusion Mixer Measurement Simplified

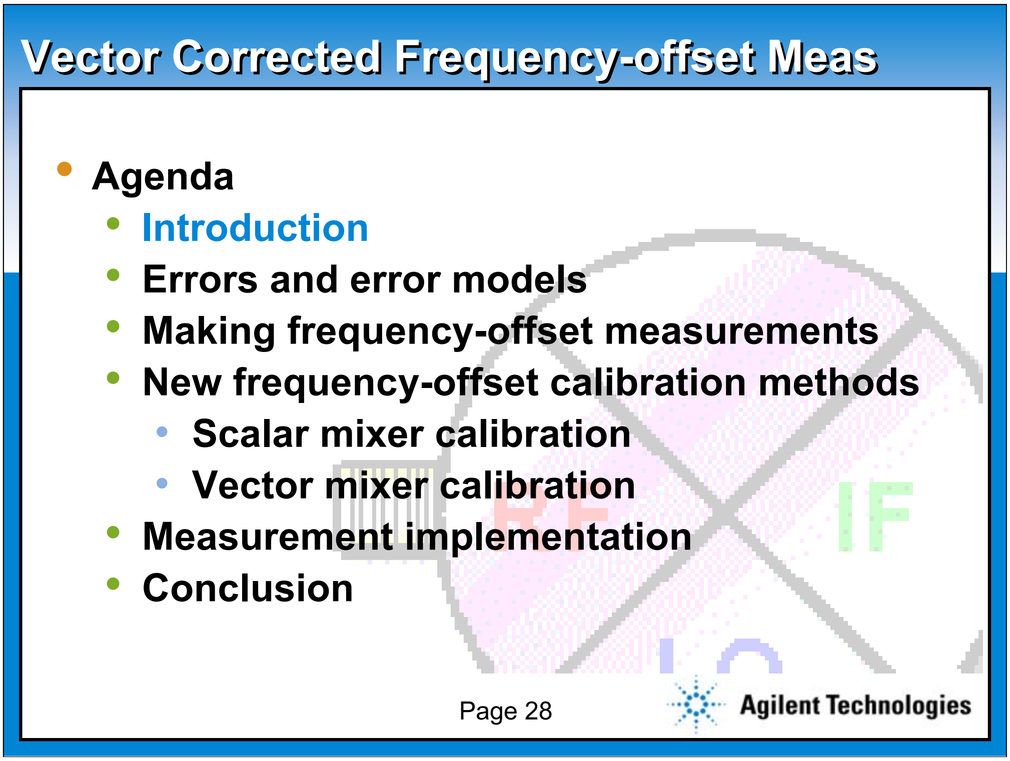
Key Measurements

- Transmission
  - Output power
  - Conversion Loss
  - Phase
  - Group Delay
- Reflection
  - VSWR / RL
- Isolation
- Distortion
  - Conversions Compression
  - Higher order mixing products
  - Two- tone IMD

Key Solutions

Q&A1

10 mins



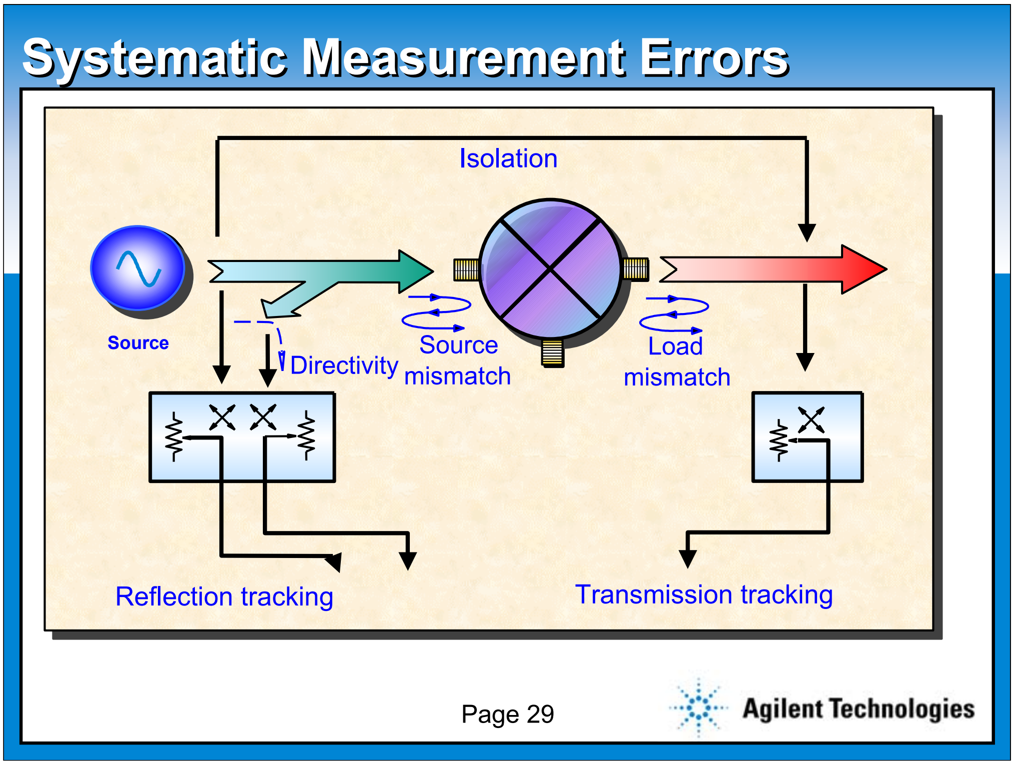
Welcome to module two of the presentation today. If you are just joining us, welcome. The first module looked at the fundamentals of frequency translation device measurements. We looked at what makes them unique and so difficult to test. We looked at the full complement of measurements needed to characterize such devices. The measurements can be grouped into transmission, reflection, isolation, and distortion measurements. And lastly we looked at several different instrument setups for making those measurements.

In module two, we will be looking specifically at using a vector network analyzers in frequency-offset mode to measure frequency translation devices.

We will introduce the meaning of frequency-offset mode and why it is required when measuring devices that have their input and output at different frequencies.

Then we will look at two new methods that have been developed by Agilent to apply vector corrected a vector error correction to the measurement of frequency translation devices. These are exciting techniques that bring the accuracy of a calibrated stimulus-response environment to the measurements of FTD. These new techniques yield a level of accuracy in both magnitude and phase measurements never before realized. We will present a few measurement examples and see how previous measurement methods compare to these new techniques.

Let’s get started.



Vector network analyzers are used to measure components because they are a stimulus response system. They have the ability characterize their own systematic error and remove them during a measurement.

There are five systematic errors that are present in network analysis measurements that are present in mixer measurements.* This diagram shows the five sources of systematic errors when making transmission measurements.

The directivity error is related to signal leakage. This will affect the accuracy of reflection measurements.

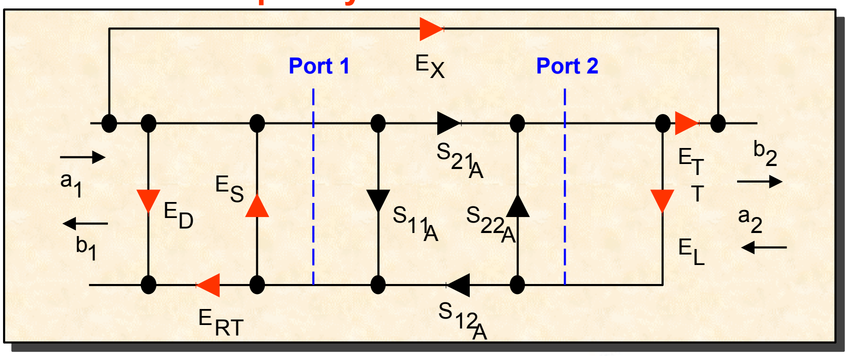
The source and load errors are related to the mismatch between the device under test and the impedance of the analyzer's measurement ports. Mixers often have poor input and output match. If the input impedance of the test system is not well matched some of the reflected signal is re-reflect into the measurement path. This re-reflection can cause uncertainties in both the magnitude and phase for reflection and transmission measurements.

Reflection and transmission tracking are related to the difference in frequency response of the analyzer's reference and measurement receivers. These errors add to the uncertainties of magnitude and phase errors for reflection and transmission measurements.

*The isolation error is omitted and assumed to be zero because the response receivers are tuned to a different frequency than the RF input.

Traditional 2-port Error Model

- Well defined linear model
- Requires
  - All 4 s-parameters be known
  - Error terms be related in phase
- Not valid for frequency translation devices



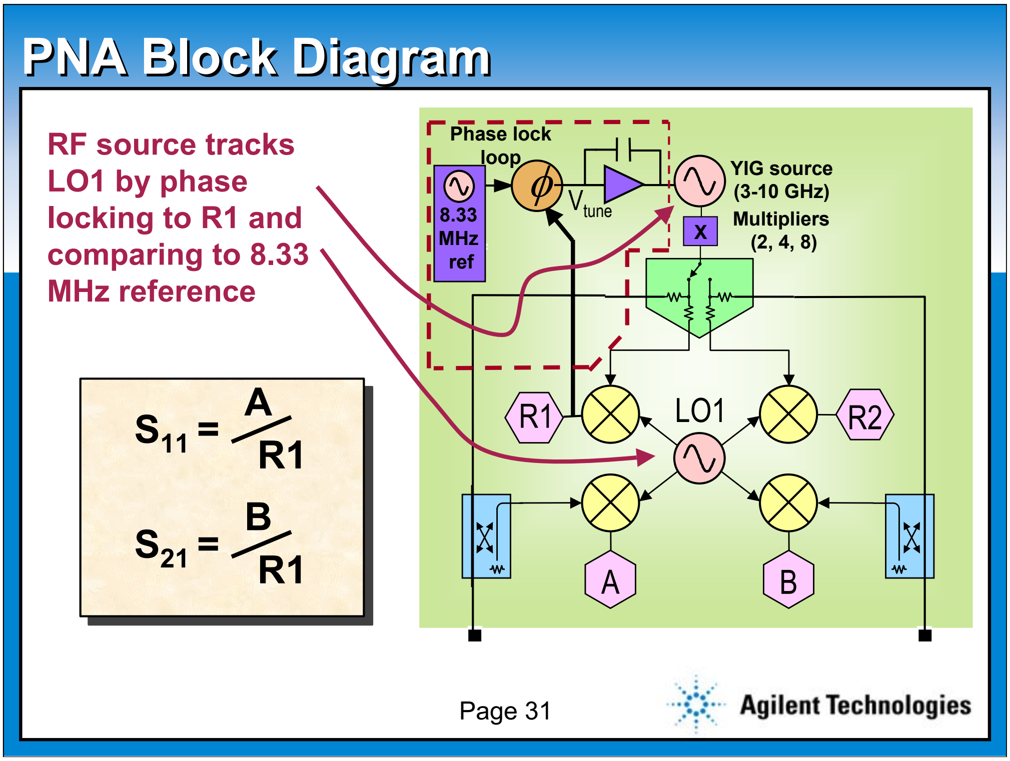
These error signals are well defined and understood. This flow graph illustrates the six terms that we saw on the last slide that are present in a transmission measurement.

Let’s look at the traditional 2-port, 12 term error model used to correct for the systematic errors present in network analyzer measurements. There are 6 error terms that affect measurements in the forward direction, and 6 error terms that affect measurements in the reverse direction. This figure shows the error terms and signals that are present in the forward direction.

In order to correct for these error signals at the time of the measurement, the contribution of the error signals must be known. This is accomplished by performing a series of measurements on standards with known responses such as a high quality open, short, and load. The deviation from the known response can be associated with a particular error signal.

At the time of the measurement, error terms in the forward and reverse directions are combined in a series of simultaneous equations with all 4 measured S-parameters data to calculate for the actual response of the device under test.

Each of the S-parameters is a vector measurement including both magnitude and phase. The definition of S-parameters depends on the linear ratio of the response over the stimulus. Since the response and stimulus frequencies are different their phases can not be related in this way. Therefore, this definition of S-parameters is no longer valid and this 2-port error model can not be used to correct for the errors in the measurements system.

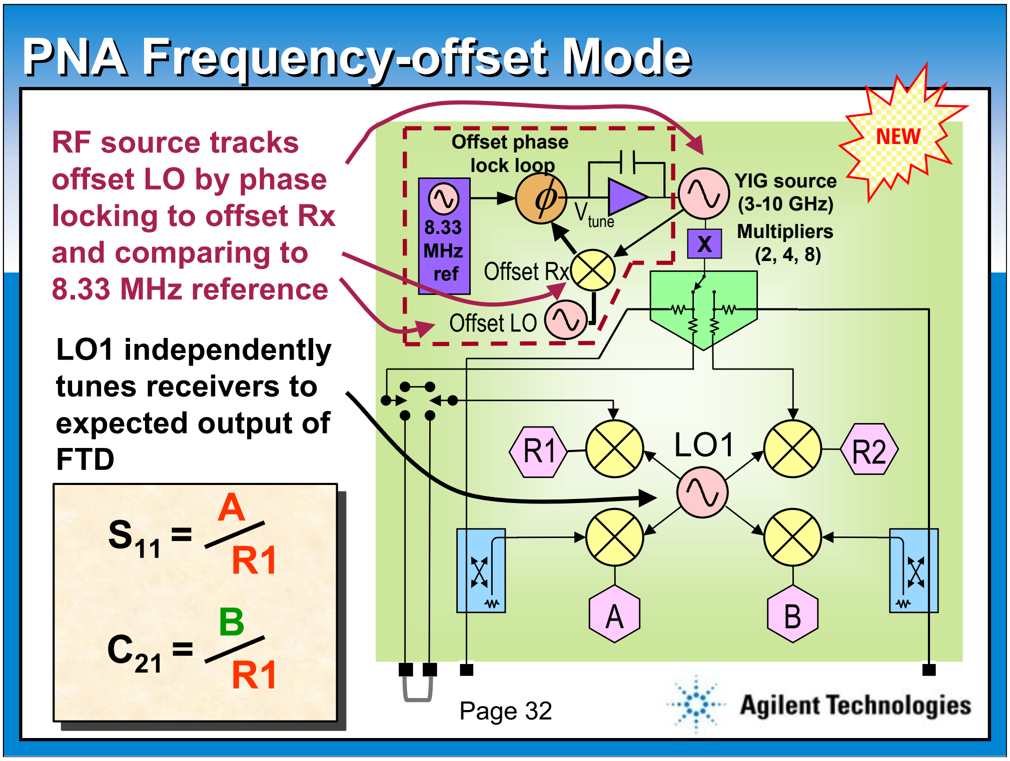


Practically this is accomplished by maintaining a very accurate phase lock between the stimulus and the response the receivers are expecting or tuned to detect. S-parameters are measured my simply ratioing the receivers and applying error correction.

When the device being measured has a non-linear response, receivers will not be tuned to this frequency.

A network analyzer produces a controlled stimulus that is presented to a device under test and the complex transmission and reflection responses can be measured against this known reference signal.

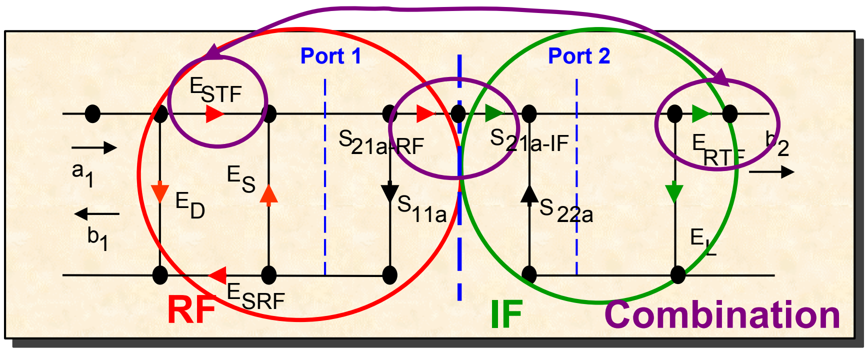
This filters out spurious signals that are not of interest so only the response of the device is detected. This is a problem for nonlinear devices that themselves produce spurious or harmonic responses. When the receivers are tuned to the input frequencies, these nonlinear responses will be out of range and will not be detected.



To accomplish this a different phase lock system is needed to enable the receivers to be set to track the expected input regardless of the stimulus frequency. A separate LO is added for the source to phase lock to that is separate from the LO used to tune the receivers. The receiver can be offset from the stimulus and thus the VNA is said to be in frequency-offset mode. Measurements are still accomplished my ratioing the receivers. But for transmission measurements the frequency being ratio are different. Linear S-parameters are no longer valid so the “S” designation can no longer be used. A new designation that that indicates that a frequency conversion has taken place during the transmission needs to be used. The “C” replace the “S” to indicate that this ratio has a different input and output frequency.

What we loose by offsetting the stimulus and receiver frequencies, is the ability to relate the phase and thus correct for the systematic error in the measurement; which is one of the major benefits of using a network analyzer in the first place.

Frequency-offset Error Model



$$E_{TF} = (E_{STF} * E_{RTF}) \qquad C_{21a} = (S_{21a\text{-}RF} * S_{21a\text{-}IF})$$

$$E_{RF} = (E_{STF} * E_{SRF})$$

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The linear systematic error model must be replaced with one that includes the frequency translation of the device under test. In this model, the error terms at the input are at a different frequency then those at the output.

It is easier to think of the model now being in two parts. One for the input at the RF frequency and one for the output at IF frequency in this example. We also need to introduce some additional terms.

The frequency translation transmission tracking error term because the composite of the source transmission tracking at the RF and the receiver transmission tracking at the IF.

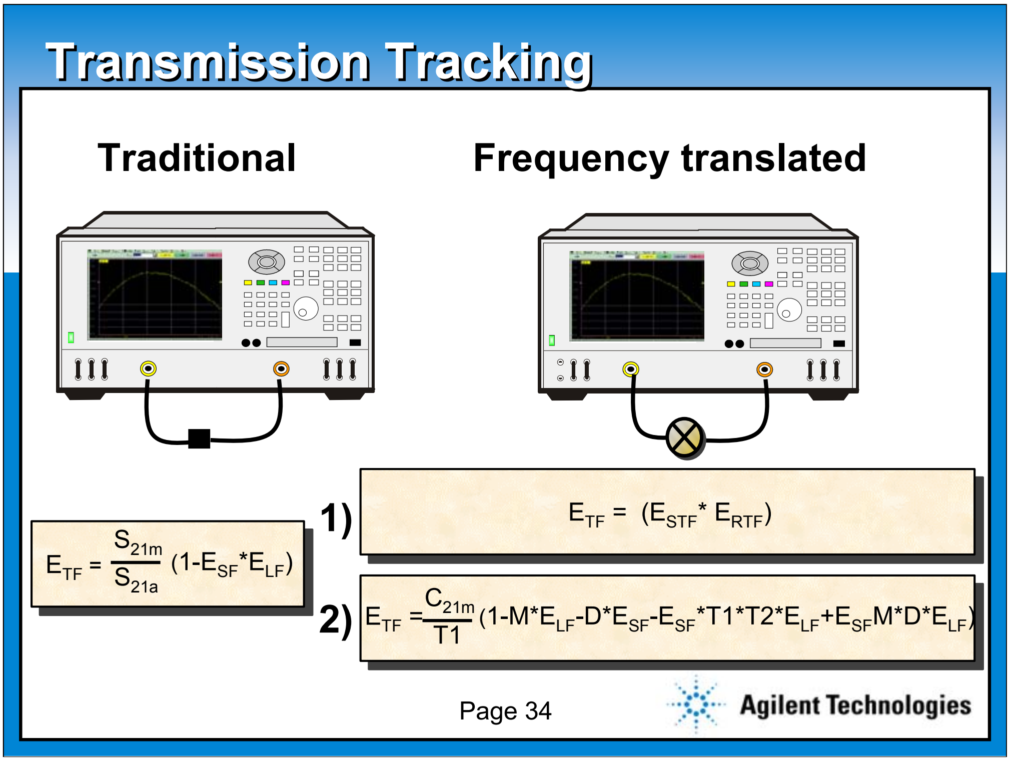
The reflection tracking is also divided into two parts now. It is the composite of the source transmission tracking and the source reflection tracking.

Lastly, the conversion measurement, C21, is divided into S21a-RF and S21a-IF to signify its two parts.

And the cross talk error is removed since it will not have an effect.

The majority of the error terms can be measured using traditional calibration means at their designated frequencies.

The frequency translated transmission tracking error term however can not. It is rather difficult to obtain. We will be looking at two different methods Agilent has developed to calculating this term. So lets take a closer look at the transmission tracking error,ETF.



In a traditional 2-port error model this is obtained by a zero length, zero loss connection in the place the device would be measured. The measured S21corrected for the source and load match is compared to the expected S21 value of zero. The difference in this ratio is the transmission tracking error. When the thru is not zero length, zero loss its characteristics must be characterized to obtain an accurate transmission tracking error measurement.

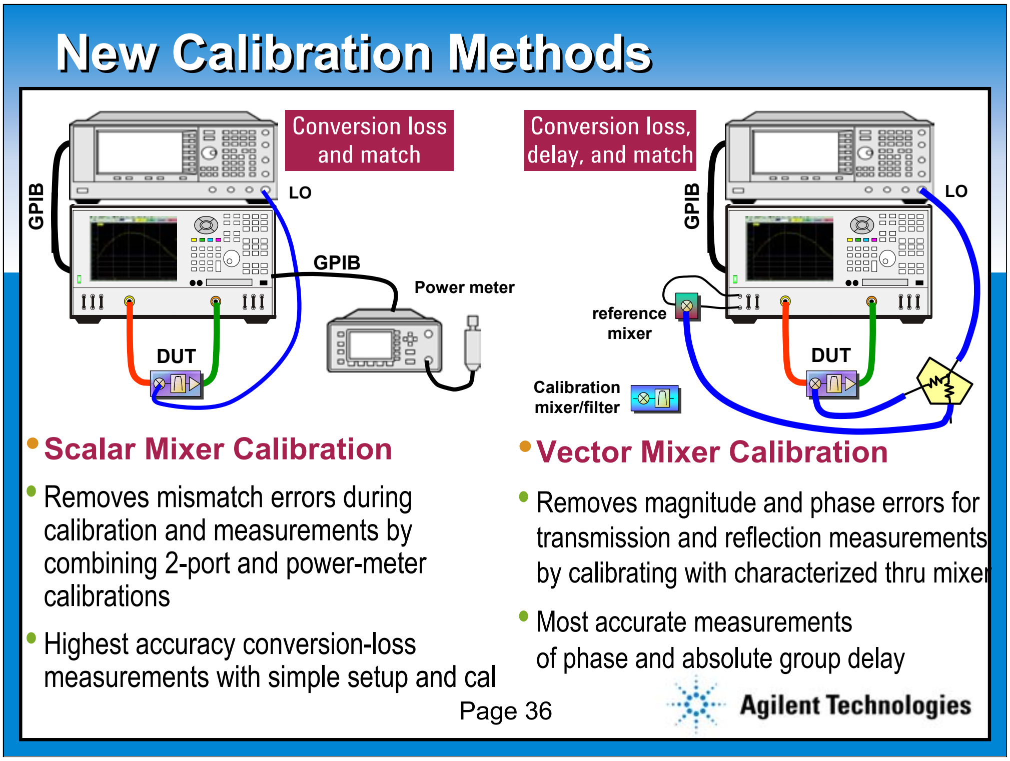
There are two methods for obtaining the frequency translated transmission tracking error term.

Method 1 measures the RF and IF portions for the error term separately and combines them to obtain a scalar transmission tracking error term that can be used in error correction.

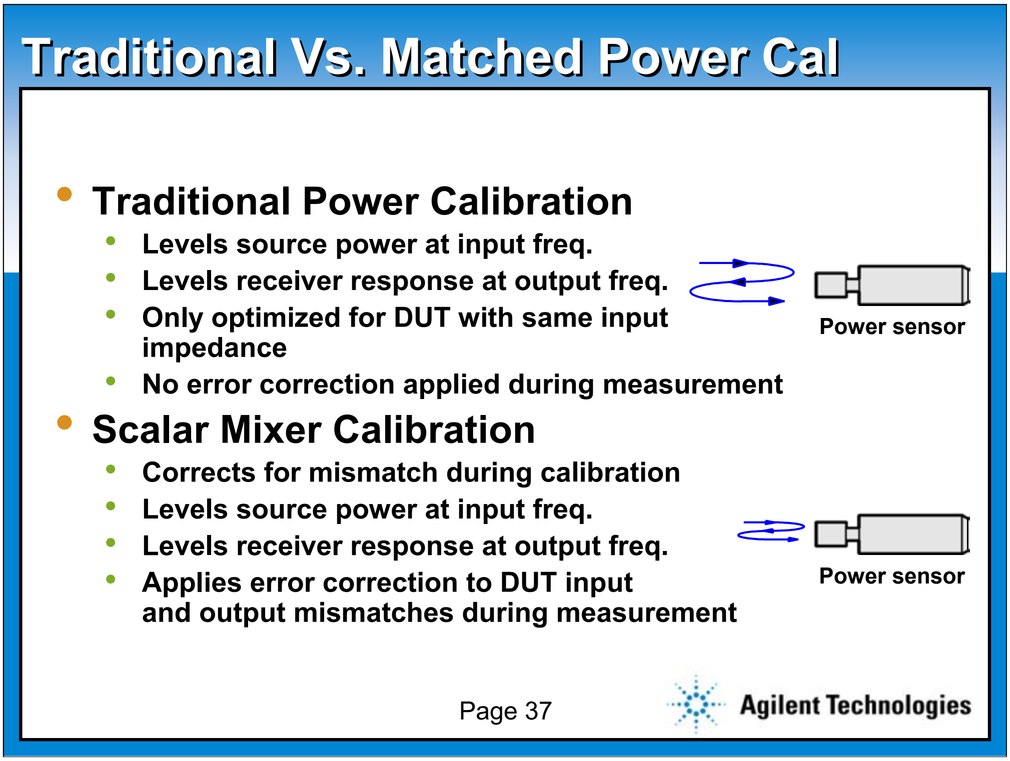
Method 2 is analogous to the characterized thru measurement. It performs and thru measurements with translation capabilities. Its conversion loss, delay, input and output match must all be known to calculated the frequency translated transmission tracking error term. Finding these terms, M, D, T1 and T2 in the above equation is that first step in the vector mixer calibration process.

Vector Corrected Frequency-offset Meas

- Agenda
  - Introduction
  - Errors and error models
  - Making frequency-offset measurements
  - New frequency-offset calibration methods
    - Scalar mixer calibration
    - vector mixer calibration
  - Measurement implementation
  - Conclusion



Let’s look first at the Scalar mixer calibration



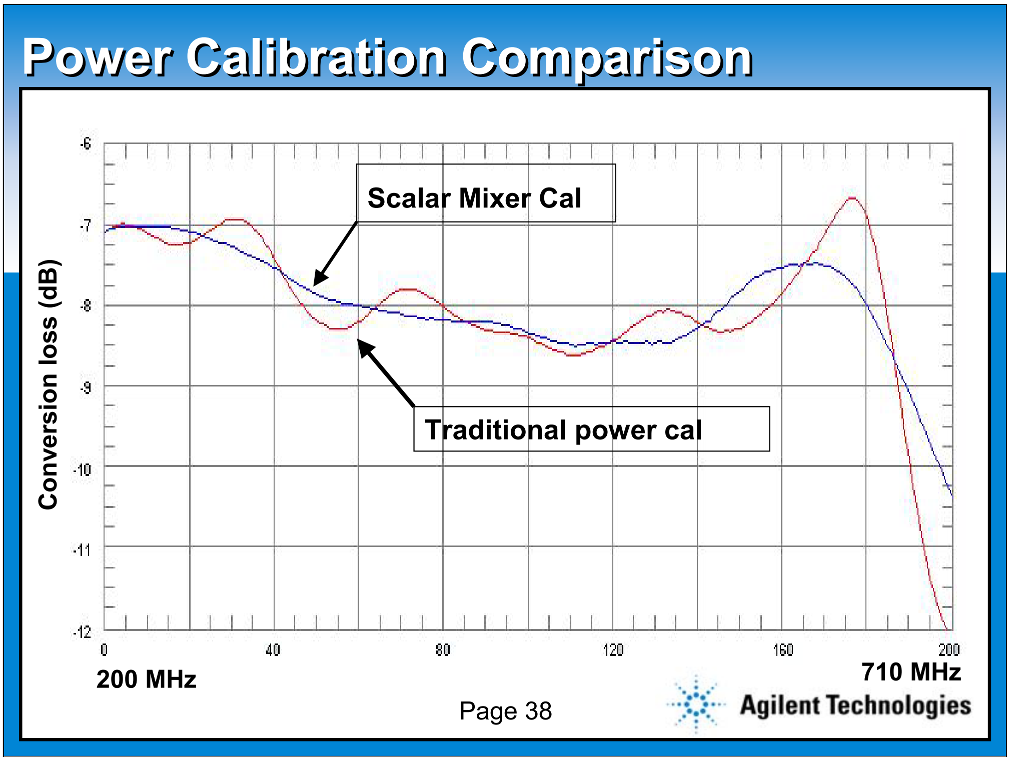
Let's look first at the Scalar mixer calibration and compare it to what is commonly done today to remove the transmission tracking error when making frequency-offset measurements. This method is to perform a source and receiver power calibration.

Since 2-port error correction is not available when making frequency-offset measurements on a network analyzer, often a source and receiver power calibration have been the best method of calibration available.

In a traditional power calibration, the analyzer source power is calibrated for flatness and linearity over the input frequency range and the receiver is calibrated for flatness and linearity over the output frequency range. The analyzer is calibrated in the presence of the impedance of the power sensor. In other words, the measurements will have optimal accuracy for devices that have the same input match as the power sensor. In this method, no error correction is applied during the measurement so any deviation between the match of the sensor and mixer will translate into measurement uncertainty.

Similarly, the match corrected calibration levels the source and receiver power at the input and output frequencies, respectively. Since there is some mismatch between the power sensor and the test cable, some (very small) power will be reflected back to the input port of the analyzer, making the power at the sensor slightly lower then it should be. Since the amount of mismatch is known for every frequency point, this can be corrected. The product is a more accurate calibration.

The other, and more important, technique that the Scalar mixer calibration performs is the error correction during the measurement itself. Using its Scalar mixer calibration error model, it is able to compensate for the input and output port mismatch between the mixer and the test system during the measurement. Without this correction, the mismatch between the mixer and the analyzer causes multiple error signals to add in and out of phase with the desired signal. This error cannot be corrected in the measurement with a traditional power meter calibration.



And what type of improvements can we see by implementing this method?

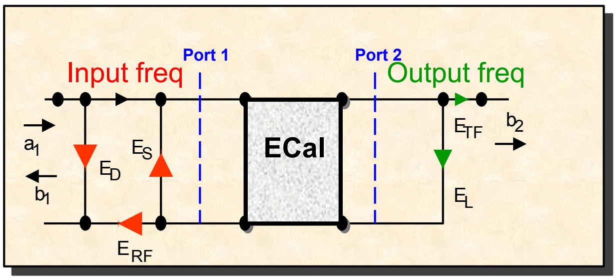
This graph shows a comparison of the two power calibration techniques measured on a test system with a poor match of –5dB. The blue (smooth) trace is a PNA measurement using the MCAC. The red trace is a measurement made with a traditional source and receiver power calibration.

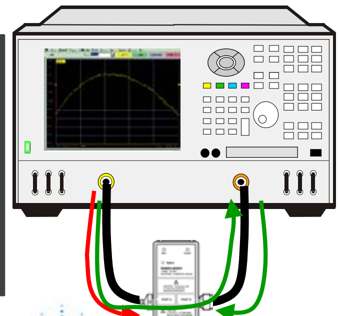
This calibration is done in two steps. Let’s take a look

Calibration Process (step 1)

- **2-port calibration at input and output frequencies**
  - **Port 1 and port 2 reference for reflection measurements**

Error term	Forward	Reverse
E_D	✓	✓
E_S	✓	✓
E_{RF}	✓	✓
E_L	✓	—
E_{TF}		—
E_X	—	—





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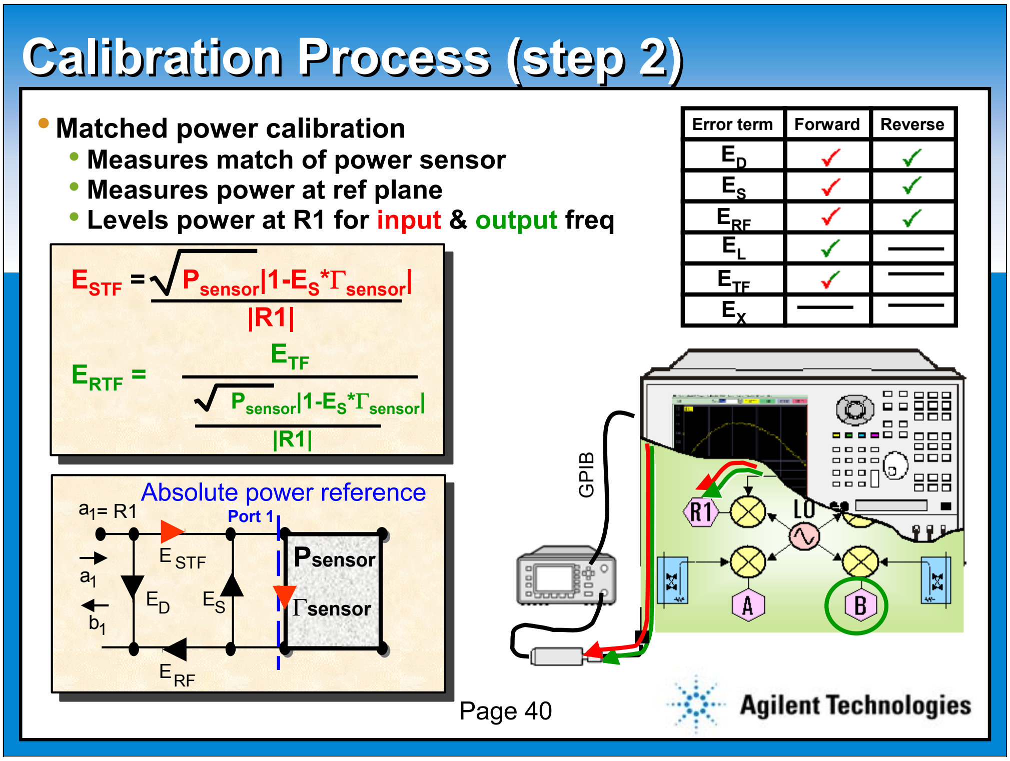
Calibration Process step 1.

The first step in performing a Scalar mixer calibration (MCAC) is a 2-port calibration over the input frequency range of the mixer. This calibration establishes a reference plane at the end of the test port cable. This is important for two reasons.

1) The directivity, source, and reflection tracking errors are determined. These error signals are needed to correct for reflection measurements on the mixer during the measurement.

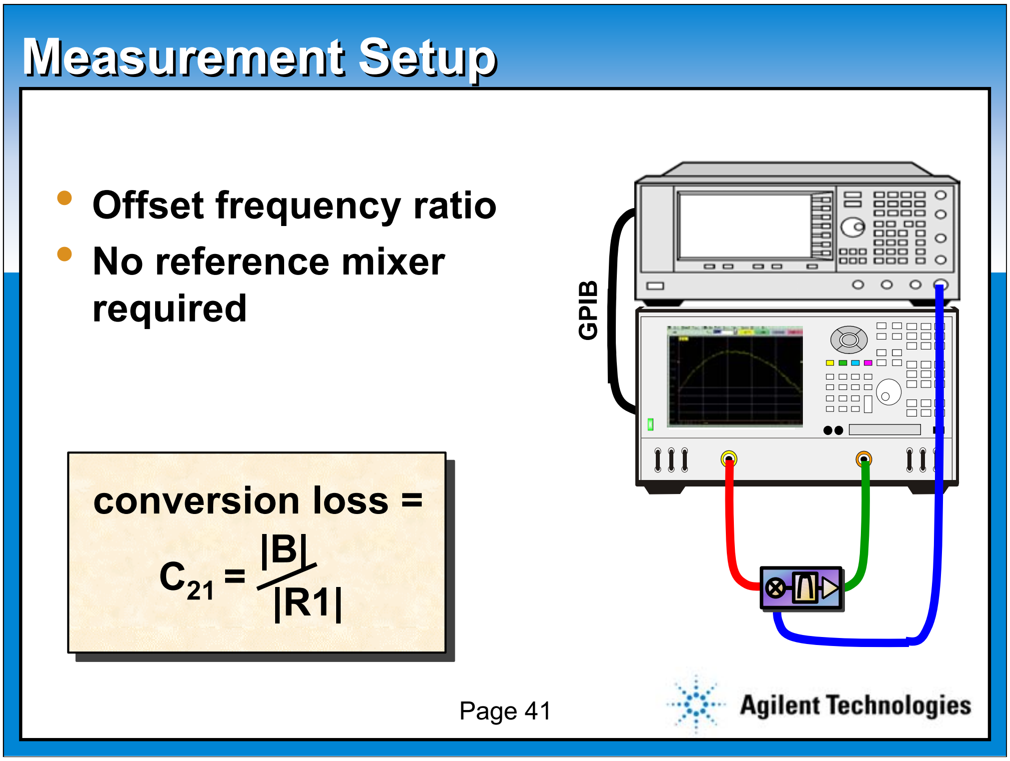
2) This calibration plane also prepares the analyzer to measure the impedance match of the power sensor in a later step when calibrating the source power. This is where the “match corrected” comes from in the Scalar mixer calibration. The match data of the power sensor is then used to improve the accuracy of the source power calibration.



Calibration Process step 2.

The second step in the process is connecting the power meter to the analyzer port 1 and performing a source power calibration. This calibration transfers the absolute accuracy of a power meter to the network analyzer for making un-ratioed measurements. The impedance of the power sensor is first measured over both the input and output frequency ranges of the mixer.

Once a power reading is made is made and the match of the sensor is known ESTF and ERTF can be solved directly from the equations given.



Now that we have looked at the calibration process, let's take a look at how simplistic the measurement setup is.

There are several things worth noting from this setup. There is no need for a mixer in the reference path that many other systems require. And there is also no filter after the mixer to filter out mixing products. Since the receiver is tuned to the correct offset, other mixing products will be filtered out and will not cause measurement errors.

To measure the conversion loss, an external source can be set at a fixed frequency while the network analyzer sweeps the input and measures the swept output. Or, the analyzer can control the external source to sweep the input and LO at the constant frequency-offset and measure a fixed output. Either way, the conversion loss will be a ratio of the input power magnitude to the output power magnitude.

This is accomplished by ratioing the B receiver to the R1 receiver.

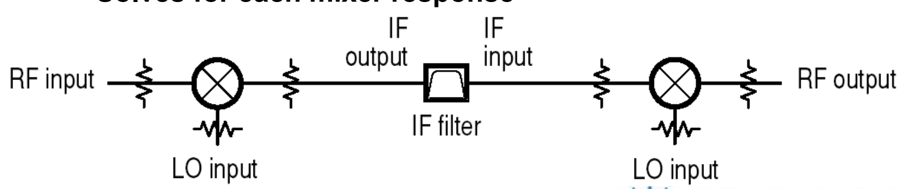
It is a ratioed measurement, but the measurements for the ratio are taken in two parts using frequency-offset mode. The B receiver is offset to the output frequency of the FTD. First a sweep in made in the forward direction and a measurement is made on the “B” receiver of the analyzer.

Vector Corrected Frequency-offset Meas

- Agenda
  - Introduction
  - Errors and error models
  - Making frequency-offset measurements
  - New frequency-offset calibration methods
    - Scalar mixer calibration
    - Vector mixer calibration
  - Measurement implementation
  - Conclusion

Current Characterization Techniques

- **Up/down Converter Method**
  - **Requires matched, reciprocal mixers ($C_{21} = C_{12}$)**
  - **Required IF filter, must remove effects**
  - **Must use synchronous (identical) LO**
  - **Does not correct for mismatch between mixers/filter/system**
  - **Attributes half to total response to each mixer**
- **Three Mixer Method**
  - **Requires three mixers, one must be reciprocal, but not match**
  - **Requires three measurements**
  - **Solves for each mixer response**



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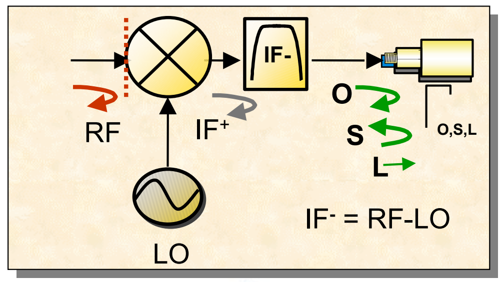
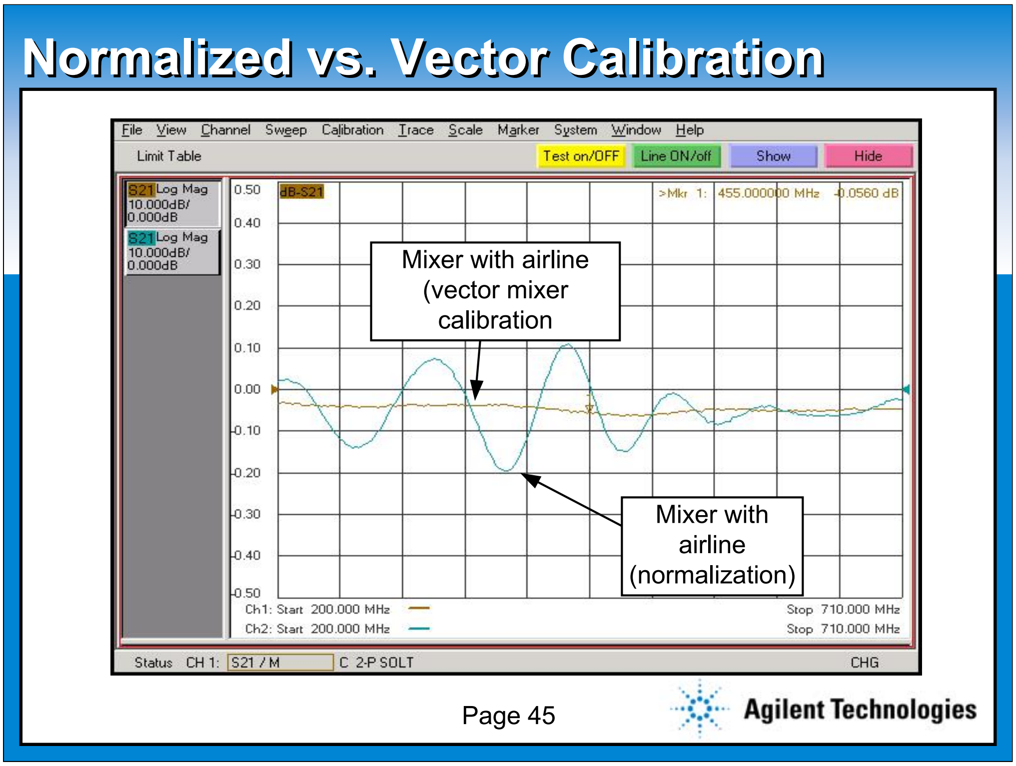
The idea of characterizing a mixer for a calibrated thru measurement is not new. There are a couple of methods that endeavor to do this .

The up/ down converter method and three mixer method are two such characterization technique.

New Characterization Technique

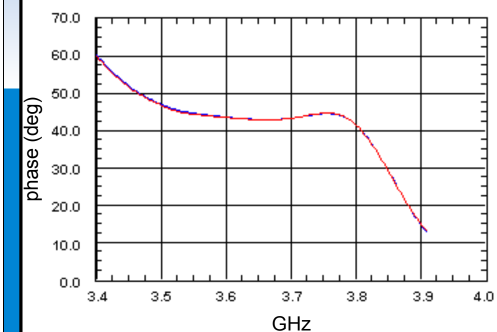
- Vector Mixer Calibration
  - Requires one reciprocal calibration mixer ($C_{21} = C_{12}$)
  - Requires an IF filter to select desired product (e.g., select RF-LO, rejecting RF+LO)
  - Assumes linear reflection at operating power
  - Synchronous LO created though self re-conversion
  - Single measurement yields complete characterization

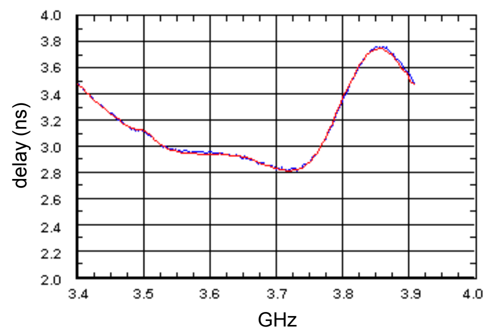




This shows the difference between a scalar calibration and the Vector mixer calibration. The error in the scalar calibration can be mapped directly to the mismatch of the mixer under test

Phase and Group Delay Measurements





Phase response of mixer (slope removed)

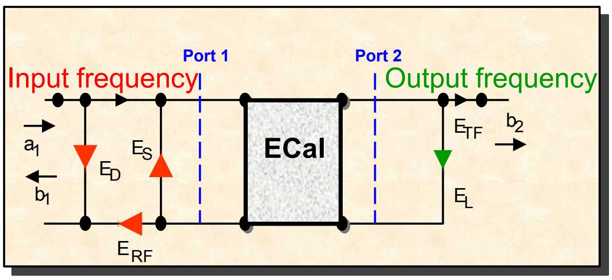
Group delay response of mixer

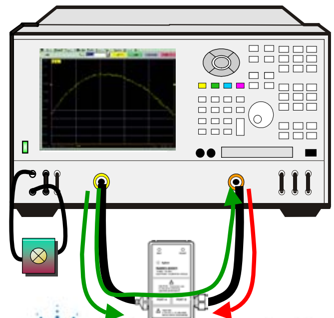
This plot shows the phase of the calibration mixer characterized as an up and a down converter. Again, there is virtually no difference. The right hand plot shows the absolute group delay of the calibration mixer (including it's image filter), and has remarkably little noise or ripple on the measurement. Depending on delay aperture, group delay resolution less than 100 ps on mixer measurements is easily obtained.

Calibration Process (step 1)

- 2-port calibration at input and output frequencies
  - Port 1 and port 2 reference for reflection measurements

Error term	Forward	Reverse
E_D	✓	✓
E_S	✓	✓
E_{RF}	✓	✓
E_L	✓	
E_{TF}		
E_X		





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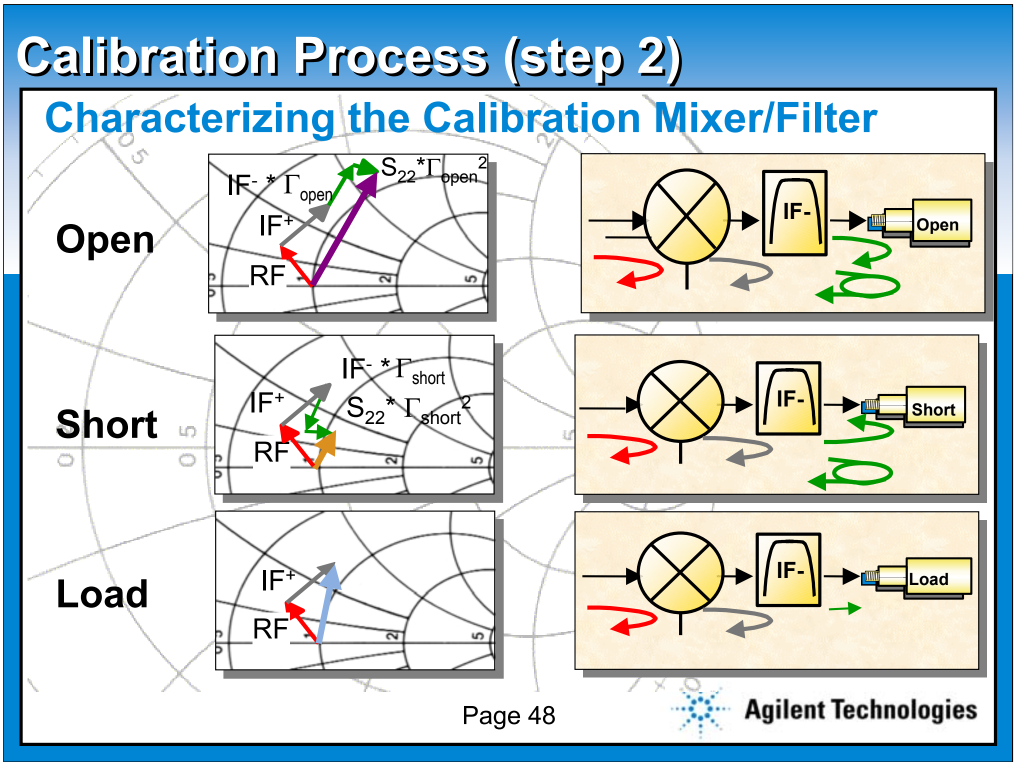
Calibration Process step 1.

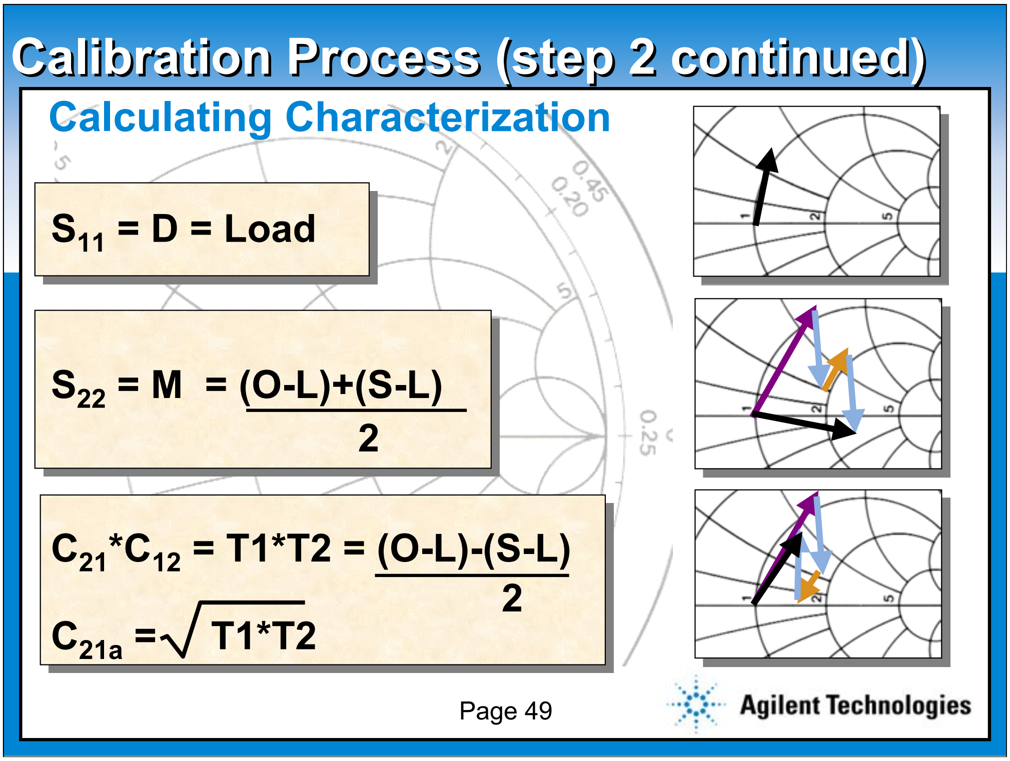
Now let’s look at how this system is calibrated. The first step is the same as the previous method, perform a 2-port calibration at the input and output frequencies. Again ECal is used to minimize the connections and simplify the calibration process.



Calibration Process step 2.

Let’s see how the mixer characterization works.

The key to this new concept is to look at a reflection measurement of a reciprocal mixer, and analyze the content of those signals. In this case, the mixer is a down converter. We identify the input signal as RF, and the output signals as IF+ to represent RF+LO and IF- to represent the desired signal RF-LO. Let’s examine what happens when we put a variety of loads on the output of the mixer. Note that the RF signal is not affected by the load, as the IF- band pass filter has a constant (high) reflection at all signals other than the IF- signal. Likewise, the IF+ signal is unaffected by the load, and it’s contribution to the the total RF signal going back to the vector network analyzer (VNA) is likewise unchanged. Note that the “IF+” signal will be reconverted by the LO to the RF frequency, and that this final RF level will include twice the conversion loss of the mixer. The “IF-” signal will pass through the filter, reflect off the load, pass back through the mixer and be up-converted to the RF frequency, again with twice the conversion loss. All of these signals add to what will be measured in a reflection (S11) measurement of the mixer, but only the portion attributable to the IF- signal will change with the load impedance. In all of these measurements, we assume the VNA has been calibrated for reflection measurements, so that there are no errors associated with the VNA port characteristics.



Now that the short, open, and load response are know for the mixer/ filter combination the input match (S11), output match (S22) and transmission response (C21) can be calculated.

Since the RF and IF+ are unchanged by the varying terminations their effects can be isolated and removed from the response of the IF- response.

There are three measurements that we need to have to calculated the frequency translated tracking error.

1) Input match, S11, called D for the calibration mixer. By definition S11 is the reflection at port one with all other ports terminated. Since none of the converted IF- signal will be reflected all the reflected signal comes from the RF and IF+ reflections.

2) The output match, S22, of the calibration mixer called M.

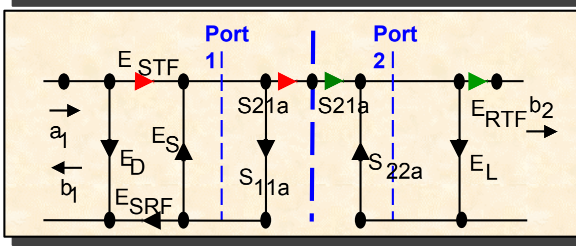
3) The one way conversion loss, C21a, of the calibration mixer. To get the conversion loss magnitude is easy, assuming S21=S12, by taking the square root the the ERF term. Getting the phase response is a little more difficult. The square root function applied to phase is really dividing the phase by 2. However, the phase wraps every 360 degrees so dividing by 2 will yield the wrong result. For example, the max phase is 180 degrees, so divided by 2 the max phase is 90 degrees, but that does not represent the real situation where the conversion phase can go to 180 degrees. Instead, what is needed is to un-wrap the phase (sometimes called expanded phase), then shift the first point phase reference to the proper offset value (and likewise shift all the other points). This shifted value is found by finding the delay (or phase slope) around the first point, and projecting it back to the zero frequency. The value at zero frequency is the offset for the phase trace. Once this is applied, the resulting trace is divided by two. Converting from the polar representation to the real and imaginary representation will re-wrap this new phase response.

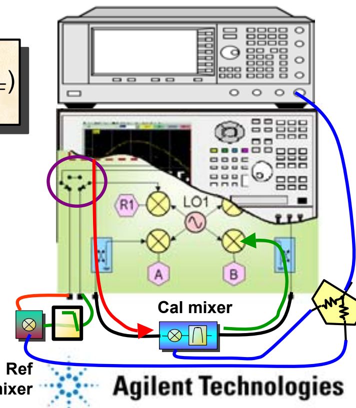
Calibration Process (step 3)

- **Create reference path**
  - **Use calibration mixer as characterized thru**
  - **Measures C_{21m} for E_{TF} calculation**
  - **Removes effects of reference mixer**

Error term	Forward	Reverse
E_D	✓	✓
E_S	✓	✓
E_{RF}	✓	✓
E_L	✓	—
E_{TF}	✓	—
E_X	—	—

$$E_{TF} = \frac{C_{21m}}{T1}(1-M*E_{LF}-D*E_{SF}-E_{SF}*T1*T2*E_{LF}+E_{SF}M*D*E_{LF})$$



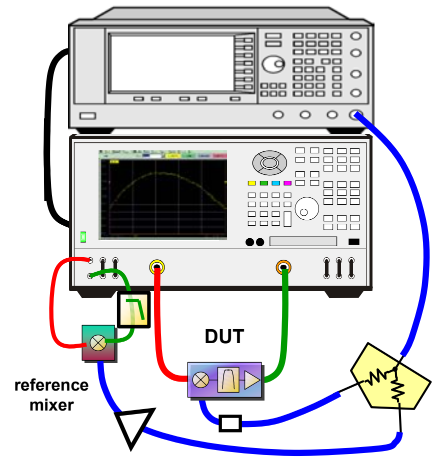


Measurement Setup

- **Ratioed measurement**
- **Reference mixer required**

conversion loss =

$$C_{21} = B / R1$$



Now that we have looked at the calibration process, let's take a look at the measurement setup.

Vector Corrected Frequency-offset Meas

- Agenda
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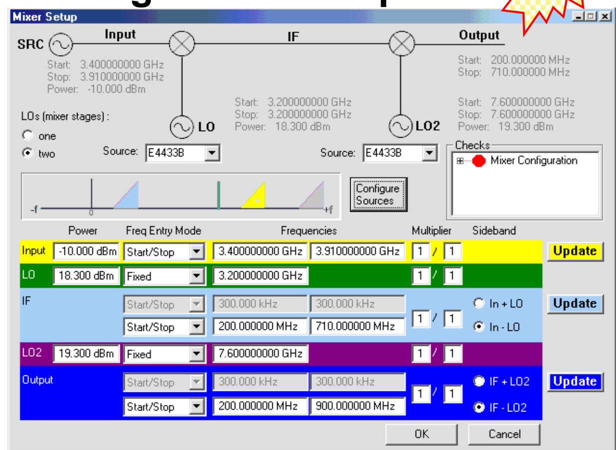
    - Vector mixer calibration
  - Measurement implementation
  - Conclusion

scalar

PNA Frequency Converter Application (FCA)

NEW

- Graphic User interface guides users
  - Measurement configuration setup
  - LO setup and control
  - Calibration process



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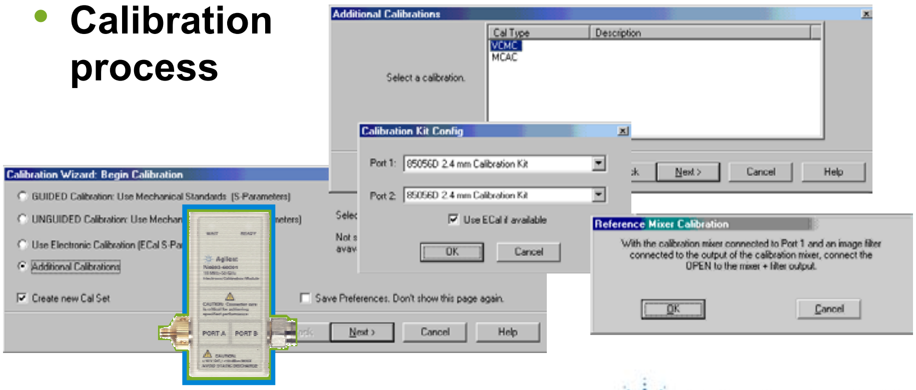
FCA Calibration Wizard

- Dialog menus guide users
  - Selecting appropriate calibration
  - Correct calibration kit (mechanical or ECal)
  - Calibration process



Conclusion

- VNA frequency-offset Mode
  - Performs all key measurements
- Systematic errors degrade meas.
- New calibration methods increase accuracy
  - Scalar mixer calibration
  - Vector mixer calibration
- PNA Frequency Convert Application
  - Implements calibration methods
  - Guides through setup and calibration

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Q&A2

10 mins