EE 117B/217B Microwave and Wireless Measurements

Lab 3 - Introduction to Spectrum Analyzers

Fall 2026

CAUTION: READ THIS PAGE CAREFULLY BEFORE STARTING THE LAB

This class uses the FieldFox Microwave Analyzer N9918B as the primary instrument. It serves as the Vector Network Analyzer (VNA) and the Spectrum Analyzer (SA). Along with the FieldFox, this class also uses several other RF instruments and equipment. Please handle the instruments very carefully as they are **very expensive** and delicate. The most vulnerable parts of the instruments are the RF PORTS (APC-3.5 connectors). These connectors are protected by the small Female-Female Port Savers, which are the silver ferrules attached to the ports. **DO NOT REMOVE THE PORT SAVERS FROM THE FIELDFOX PORTS FOR ANY EXPERIMENT.**

All connections are to be made through these Port Saver Connectors. If you have never handled RF instrumentation before, ask the instructor to demonstrate how to make connections and handle these connectors and cables. Always hold the cable with one hand (non-dominant) and rotate the end nuts of the SMA cables with the other (dominant) hand. Once you make a soft connection by hand, use the torque wrench provided to complete the connection. Note that whenever SMA connections are made, you must only rotate the end nuts and not the components; this ensures the internal pin enters and exits the jack in a straight trajectory. As far as pushing the control knobs and keys on the instrument are concerned, no special care is needed (they are fail-safe). The worst that can happen is that the instrument may lock up. In that case, press the `Preset` key or ask the instructor for help.

Caution for this particular lab

This lab uses signal generators and an RF amplifier that can deliver enough power to damage the FieldFox. NEVER APPLY MORE THAN +20 dBm OR ANY DC VOLTAGE TO THE RF INPUT PORT OF THE FIELDFOX (Port 2). When raising the source power, step it up gradually and double-check the minus sign before entering a value. Keep the 10 dB attenuator at the analyzer input whenever the amplifier is in the signal path, check the DC supply settings before connecting the amplifier, and never exceed the maximum input ratings of the amplifier and mixers given in their datasheets.

Introduction

In Labs 1 and 2 we used the FieldFox as a Vector Network Analyzer to measure the S-parameters of passive devices, first in the frequency domain and then in the time domain. In this lab we will be using the **Keysight FieldFox N9918B Handheld Microwave Analyzer** in Spectrum Analyzer (SA) mode. The FieldFox N9918B measures signals directly from 30 kHz to 26.5 GHz. Unlike the VNA, which measures the response of a device to a stimulus it supplies itself, the SA measures the power spectrum of whatever signal arrives at its input. This makes it the primary instrument for characterizing signal sources and the *nonlinear* behavior of active devices — harmonics, gain compression, intermodulation and frequency conversion — none of which appear in the linear S-parameters measured in the previous labs.

The lab begins with the settings that determine what the analyzer can resolve and how weak a signal it can see: the resolution bandwidth, the video bandwidth, the number of points, and the sweep type. These are then used to measure the harmonic distortion of a signal source, and the 1 dB compression point and third-order intercept of an RF amplifier. The last part extends the analyzer beyond 26.5 GHz with an external harmonic mixer module, to measure a 72 GHz millimeter-wave source.

Objectives

Building on the linear S-parameter measurements of Labs 1 and 2, the purpose of this lab is to introduce the spectrum analyzer and the measurements of signal sources and nonlinear devices that it enables. At the end of this lab you will:

1. Understand how the resolution bandwidth, video bandwidth, number of points, and sweep type (Step vs. FFT) set the frequency resolution, noise floor, and speed of a spectrum measurement.

2. Measure the harmonic distortion of a signal source, and verify that the harmonics you observe come from the source and not from the analyzer itself.

3. Measure the 1 dB compression point of an RF amplifier and, using a two-tone test, its third-order intercept point (IIP3), after first establishing the distortion floor of the test setup.

4. Extend the frequency coverage of the analyzer beyond 26.5 GHz with an external harmonic mixer module, plan the LO and IF frequencies yourself, and identify the image and harmonic spur responses of an unpreselected mixer.

This lab is divided into four parts: (1) Resolving closely spaced signals, (2) Harmonic distortion, (3) 1 dB compression and IIP3 of an amplifier, and (4) External millimeter-wave mixers.
Since 3 students are performing the lab together, please ensure that you **take turns on performing the experiments.**

Equipment List

Table 1 and Fig. 1 list the instruments, components, and devices under test used throughout this lab. Verify that all items are present at your bench and in your Lab Box before beginning, and consult the instructor if anything is missing or damaged. Start by reading the datasheets of the components (in particular the amplifier, the mixer, and the millimeter-wave modules); you will need their specifications and maximum ratings throughout the lab.

Lab Report

Each lab comes with a Lab Questionnaire. As you go through this Lab Manual, you will encounter several questions along the way. When you do, go over to that question in the Lab Questionnaire and answer it accordingly. The list of questions and answers will be turned in as your Lab Report. All students in one lab group will submit one combined lab manual.

Equipment	**Part Number**	**Qty**
Keysight FieldFox Microwave Analyzer	N9918B	1
RF signal generator S2	N5181A	1
SMA coaxial cables	NA	3
Female-Female SMA adapters (barrels)	NA	2
4-way power combiner	ZC4PD-153-S+	1
10-dB attenuator (pad)	FW-10A+	1
1 GHz low-pass filter (LPF)	SLP-1200+	1
RF amplifier	ZX60-123LN-S+	1
DC power supply	NA	1
50 Ω SMA terminations	NA	1
VDI PSAX spectrum analyzer extension module (WR-12, 60–90 GHz)	PSAX	1 (shared)
VDI PSGX source extension module (72 GHz transmit)	PSGX	1 (shared)
Lock-in amplifier (Procedure C)	SR830	1
Torque wrench	NA	1
USB flash drive	Sandisk	1
PC workstation with MATLAB / Python	NA	1

Table 1: Equipment used in the Spectrum Analyzer Lab 3 experiments.

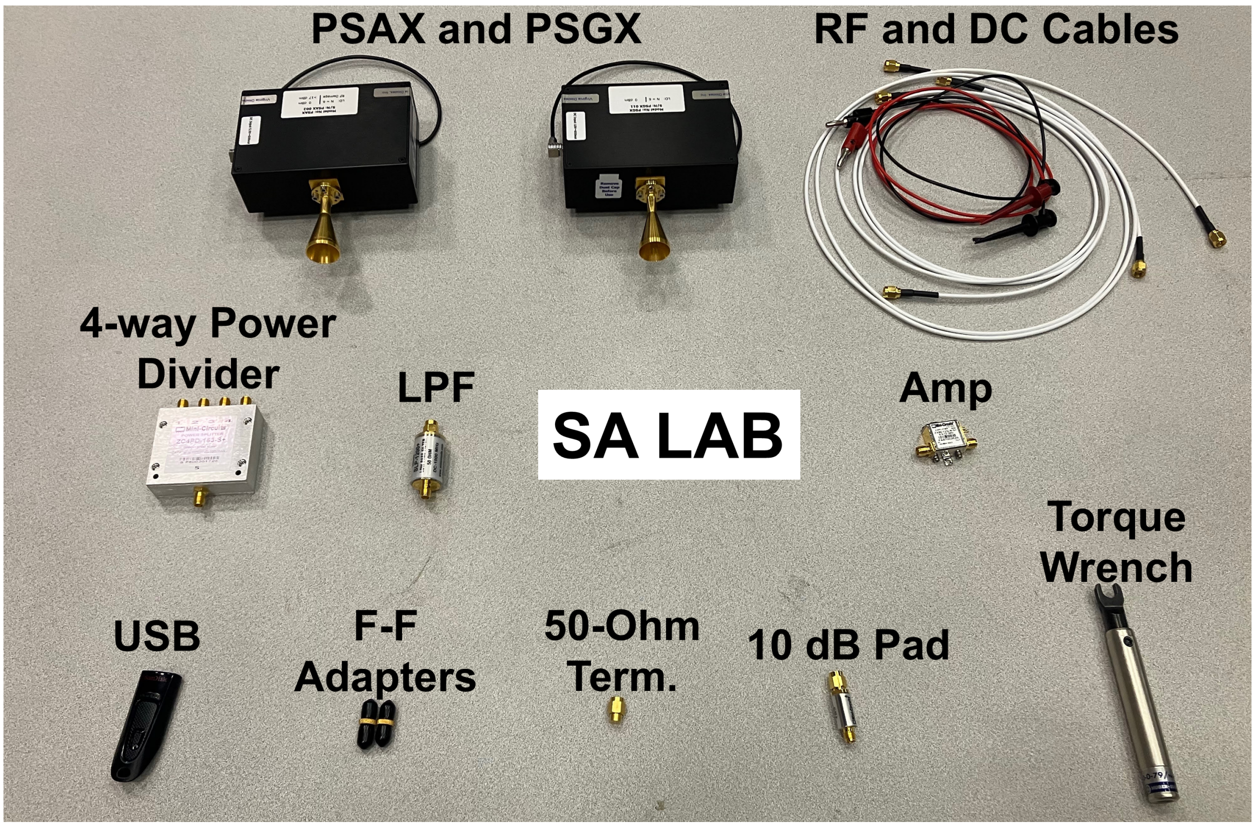

Figure 1: Components used in this lab (SA).

1 Resolving Closely Spaced Signals

What Is Resolution Bandwidth? Signal resolution is determined by the intermediate frequency (IF) filter bandwidth. The spectrum analyzer traces the shape of its IF filter as it tunes past a signal. Thus, if two equal-amplitude signals are close enough in frequency, the filter shapes for the two signals can fall on top of one another and appear as a single response. If two signals are not equal in amplitude but are still close together, the smaller signal can be hidden under the response of the larger one.

Spectrum Analyzer Function Used. The resolution bandwidth function (`Res BW`, RBW) selects the appropriate IF bandwidth for a measurement. (Resolution bandwidth is specified as the 3 dB bandwidth of a filter.) The following guidelines can help you determine the appropriate resolution bandwidth to choose.

1.1 Characterizing and Filtering the FieldFox Internal Source

For many of the experiments in this lab we will use Port 1 of the FF as a signal generator. This lets a single instrument both provide a tone and analyze the spectrum. The internal source is convenient but contains many harmonics since it is closer to a square wave, so we characterize and clean it before using it.

1. Turn on the FieldFox and select Spectrum Analyzer mode: `Mode` → `SA`. In SA mode, press `Measure` → `Source` and set the `CW Freq` to 1 GHz with a `Source Power` of −10 dBm. Connect the source output (Port 1) through an SMA cable to the RF INPUT (Port 2). Turn on the Source. Notice the large number of harmonics.

2. Set the Span to 0.9–1.1 GHz: `Freq/Dist` → `Start 0.9 GHz` and `Stop 1.1 GHz`. Now increase the span so that it covers 0.9–15 GHz: you will see a comb of harmonics of the 1 GHz tone. This is because the internal source from Port 1 cannot produce a pure CW signal. Turn off the source.

3. To remove the harmonics, attach the 1 GHz LPF between Port 1 and the SMA cable. Turn on the source and check the spectrum again over 0.9–1.1 and 0.9–15 GHz; the harmonics above 1 GHz should now be strongly attenuated up to a certain frequency. Reduce the Resolution Bandwidth (RBW) to lower the noise floor: `BW` → `Res BW Enable` → `10 kHz`. What is the strongest harmonic you see and what is its frequency (**Q1.1.1**)? Why do the harmonics at higher frequencies not get attenuated as much (**Q1.1.2**)? (Hint: check the S_{21} of the filter by switching to the VNA mode.) **We will use this internal source as source S2 in future measurements.** Turn off the internal source.

1.2 Stepping Through a Measurement of Two Equal Amplitude Signals

Generally, to resolve two signals of equal amplitude, the resolution bandwidth must be less than or equal to the frequency separation of the two signals. For example, to resolve two signals of equal amplitude with a frequency separation of 1 kHz, a resolution bandwidth of 1 kHz or less should be used.

1. Set the Center frequency to 1 GHz and Span to 10 MHz. Reduce RBW to 10 kHz. Connect the two signal sources (S1 and S2) through the power combiner to the FieldFox RF INPUT

(Port 2). Refer to the Equipment list table for the Source and combiner information. Source S1 is the Vector Signal Generator (VSG) and S2 is the FieldFox (FF). You can choose any 2 of the four input ports of the 4 way power combiner. Set the signal power and frequency of source S1 to −20 dBm and 999.4 MHz, respectively. Leave the output RF OFF.

2. On Source S2 set the frequency to 1000.6 MHz and amplitude to −20 dBm. **Double check to make sure the power levels are correct** and turn on the outputs of both S1 and S2. Place markers on the two peaks and adjust the output signal amplitude of S1 (VSG) such that the marker reading on the spectrum analyzer is the same as for S2 (FF). Now signals generated by both sources S1 and S2 are of equal magnitude (at the SA port) and are separated by 1.2 MHz in frequency. What is the amplitude reading on the two sources, and what is the amplitude of the two peaks on the SA (**Q1.2.1**)?

3. Set the resolution bandwidth to 100 kHz and the video bandwidth to 10 kHz:

   `BW` → `Res BW` → 100 kHz
   → `Video BW` → 10 kHz

   Plot the results (**Q1.2.2**).

4. Change the resolution bandwidth to 3 MHz and notice that you can no longer see that there are two signals present. Increase the span to 20 MHz. Plot the results (**Q1.2.3**).

5. To resolve two signals with a frequency separation of 1.2 MHz, an RBW of less than 1.2 MHz must be used. Use the maximum RBW that still resolves the signals (the dip needs to be at least 3 dB below the peaks). Plot the results. What is the resulting RBW you obtained (**Q1.2.4**)?

6. Increase the `Video BW` to 1 MHz. Notice that the dip at 1 GHz is unstable. Can you explain why (**Q1.2.5**)?

7. Select `Trace` → `State` and change the states between `MaxHold` and `MinHold`. During `MaxHold` the dip disappears and during `MinHold` the dip is very low. During the `MaxHold` measurement note down the peak value. Can you explain why (**Q1.2.6**)? This also serves as a hint for the previous question.

8. Restore `Trace State` to `Clr/Wr` and reduce the RBW to 100 kHz, VBW to 100 kHz and increase the span to 100 MHz. Now we will change another setting in the SA which is the number of points. Press `Sweep` → `Points`. Set this to 101. Now you are again in a situation where both signals are partially intersecting. But note that there is no fluctuation in the dip even when VBW is set to 1 MHz. Can you explain why (**Q1.2.7**)?

9. Change the number of points to 1001. Clearly now the two peaks are separated. Note that in this case the sweep time is still the same. But this is only the case when the SA operates in FFT mode (shown in the bottom left corner of the screen). The typical mode of operation of the SA is called the STEP mode in which case the sweep time is a function of the number of points. You can manually toggle between them using `BW` → `Advanced` → `Sweep Type`. We will explore these two modes in more detail next. Before proceeding return the `Sweep Type` to `Auto`.

10. Turn off both sources and connect source S1 directly to the SA. After making the connection, set the frequency of S1 to 1 GHz and amplitude to −20 dBm and turn `RF On`. On the SA,

press `Preset` and `Mode Preset` (Factory). Set the SA center frequency to 1 GHz, span to 100 MHz and RBW to 1 MHz. The SA operates in two modes: STEP and FFT. Notice that the mode is currently Step as shown in the bottom left side of the screen. Reduce the RBW in steps and find the transition point at which the SA switches from Step to FFT mode (**Q1.2.8**). It automatically switches between them depending on the span and the RBW. The difference between the two modes is discussed in the lecture.

11. Reduce the VBW to 1 kHz and set the RBW to 1 MHz. Slowly reduce the RBW from 1 MHz down to 1 Hz, one decade at a time. Notice the noise floor drops. Can you explain why? Does the drop in the noise floor per decade meet your expectations (check between Step and FFT modes) (**Q1.2.9**)?

EE 217B: Shape Factor of the IF Filter
Measure the shape factor of the IF filter in both STEP and FFT modes. It is the ratio of the 60 dB to the 3 dB bandwidth. Set the Span, RBW and VBW accordingly. If you need to increase the source power, do so carefully and make sure not to exceed 0 dBm. What is the shape factor in STEP and FFT modes (**Q1.2.10**)?

Turn off the Source S1 (RF OFF).
Reminder: Take turns in doing the experiments.

2 Harmonic Distortion

What Is Harmonic Distortion? Most transmitting devices and signal sources contain harmonics. Measuring the harmonic content of such sources is frequently required. In fact, measuring harmonic distortion is one of the most common uses of a spectrum analyzer. Harmonic distortion can be checked very quickly by using the measurement routine described below. This measurement routine measures harmonic amplitudes relative to the source (fundamental) frequency.

There are a few ways to measure harmonic distortion using a spectrum analyzer. Procedure A illustrates the fastest method, which permits a simultaneous display of the fundamental and its harmonics, but it is also the least accurate. Procedure B takes longer to perform but provides a better measurement of harmonics closer to the noise floor. Procedure C is useful when measuring very low harmonic distortion sources. The equations used to convert the measured harmonic levels into percent distortion are available in Appendix A.

2.1 Fast Harmonic Measurement: Procedure A

This example measures the harmonic content of a 300 MHz signal provided by a signal generator.

1. Set the signal power level of source S1 to -10 dBm and frequency to 300 MHz. Connect the signal source to the spectrum analyzer RF INPUT (50 Ω). Press `Preset` → `Mode Preset` to start the procedure from a preset state. Turn on Source S1.

2. Set the start frequency to 200 MHz, the stop frequency to 1 GHz and the RBW to 1 kHz. This should display the fundamental frequency, the second and third harmonic.

3. The first question we should answer is whether these harmonics are coming from the source or generated internally by the SA? Recall that the first stage of the SA is an attenuator and

the second stage is a mixer, the latter of which can be driven to non-linearity. To verify this, we can change the attenuator value and note any changes in the harmonic levels.

4. Set a marker at 300 MHz, a second marker at 600 MHz and a third marker at 900 MHz. Turn on `Marker Table`. Press `Scale/Amptd` → `RF Attenuation` and change it to `Manual`. Note down the marker values and increase the attenuation from 15 dB to 25 dB. If there were no changes to the marker levels, it means the mixer is not generating harmonics of its own. Why (**Q2.1.1**)? Can you explain why the noise floor apparently went up when the attenuation value was changed (**Q2.1.2**)?

5. For measurement accuracy, raise the peak of the fundamental to the reference level: select the first marker, then `Mkr/Tools` → `Marker→Ref Level`. Plot the resulting display (**Q2.1.3**).

6. **Find the harmonic distortion (Method 1).** Use the marker functions to measure the difference in amplitude between the fundamental and the second harmonic. Use Eq. (5) (Appendix A) to determine the Total Harmonic Distortion (THD) due to the presence of the second harmonic. Repeat for both the third and fourth (if visible) harmonics. Also indicate the S/N for each measurement (**Q2.1.4**).

7. **Measure additional harmonics.** Repeat the same procedure using source S1 with the same signal specifications. Extend the stop frequency to see two more harmonics than in the previous case. Note that to lower the noise floor you can set the RF attenuation to 0 dB. But make sure the marker levels are the same (i.e. no harmonics from the mixer). In order to use the `Next Peak` function (in `Mkr/Tools`) for the smaller peaks, you may need to lower the `Peak Threshold` setting which is in `Mkr/Tools` → `More` → `Mkr Properties` → `Peak Criteria`. Report the harmonic distortion from each harmonic. What differences do you observe between the two cases (3 harmonics vs. 5 harmonics cases) in the Total Harmonic Distortion (THD)? Can you explain this (**Q2.1.5**)?

8. **Find the harmonic distortion (Method 2).** Another easy way of determining the percent of distortion is to change the amplitude units to volts:

   (a) Press `Scale/Amptd` → `More` → `Unit` → `V`. The marker readout automatically changes to voltage units.

   (b) To determine the percentage of distortion, use Eq. (4) (Appendix A) (**Q2.1.6**).

2.2 Accurate Harmonic Measurement: Procedure B

This method is a little longer, but because each signal is measured in a narrower span and resolution bandwidth, the signal-to-noise ratio is improved, making the results more accurate.

1. Using the present setup, clear the markers from the screen (`Marker` → `More` → `Markers All Off`) and restore the amplitude units to dBm.

2. **Measure the fundamental.** Reduce the frequency span:

   (a) Press `Marker` → `Peak`, which activates a marker on the highest-level signal on screen.

   (b) Move the signal to the center of the display (`Mkr/Tools` → `Mkr→Center`).

   (c) Reduce the frequency span to 1 MHz. You may need to correct the center frequency value.

3. Set the center frequency step size to the frequency of the fundamental so that stepping the center frequency moves by exactly one harmonic:

   `Freq/Dist` → `More` → `CF Step` → 300 MHz

4. For better amplitude accuracy, set the marker to the peak and move the signal to the top of the graticule: `Mkr/Tools` → `Marker`→`Ref Level`. Plot the resulting display (**Q2.2.1**).

5. **Measure the second harmonic.** Step the center frequency up by one step (which equals the fundamental frequency = 300 MHz). Select Center frequency and use the step-up (⇑) button. This sets the spectrum analyzer center frequency to the second harmonic. Use the Signal Track feature to automatically track the peak with the marker: `Mkr/Tools` → `More` → `Signal Track ON`. Move `Mkr`→`RefLev` if needed.

6. **Find the harmonic distortion.** Use Eq. (5) (Appendix A) to convert the difference between the second harmonic and the fundamental to a percentage of distortion. Again, units can be changed to volts in order to read the voltage ratio of the two signals.

7. **Measure additional harmonics.** For each additional harmonic you wish to measure, press the step-up (⇑) key for the center frequency and move `Mkr`→`RefLev` as needed. Starting from the fourth harmonic onward, reduce the RBW and Span to get a stable measurement. Find the harmonic distortion for all harmonics up to the sixth. You may need to turn off Signal Track at times, especially when the next harmonic is below the noise floor. Also report the SNR in dB for each measurement. Report the Total Harmonic Distortion (THD). Compare with the values found using Procedure A (**Q2.2.2**).

8. **Percent of harmonic distortion.** Using the amplitudes of the harmonics in volts, compute the total percent of harmonic distortion as described in Appendix A (**Q2.2.3**).

9. Turn off the source S1 (RF OFF).

2.3 Very Accurate Harmonic Measurement at Lower Frequencies: Procedure C

EE 217B: Harmonic Distortion with a Lock-In Amplifier
For extremely low THD signal sources, the procedures described above cannot be used since the THD of the SA is not low enough. In these cases there is an alternative experimental setup which makes use of a different instrument called a Lock-In Amplifier. Despite the name, this instrument is really closer to an IQ Mixer. It measures the amplitude of a signal and its phase relative to a reference signal. It can measure ultra weak CW signals (down to −150 dBm) even in the presence of noise. Ask the instructor for the setup and the measurement procedure. Measure the harmonic distortion of the source with the lock-in amplifier, report the level of each harmonic you can detect and the resulting THD, and compare the weakest harmonic you can measure this way with the weakest one Procedure B could resolve on the spectrum analyzer. Where does the difference come from (**Q2.3.1**)?

Reminder: Take turns in doing the experiments.

3 1 dB Compression and IIP3

Every real amplifier or active device is linear only over a limited range of input powers. As the input power increases, the device eventually saturates and its gain drops. Two figures of merit are used to quantify this behavior: the **1 dB compression point** (a single-tone measure of gain saturation) and the **third-order intercept point** (a two-tone measure of distortion generation). In this section you will measure both for an RF amplifier using the FieldFox in SA mode together with the available signal generators. For the two-tone intercept measurement, one of the two tones is supplied by the FieldFox's own internal source and the other by the external generator S1, similar to the RBW experiments we did in the beginning of the lab. The background theory for both measurements is covered in the lecture notes on nonlinearity.

3.1 Part A: Small-Signal Gain

1. Press `Preset` on the FieldFox and select SA mode. Connect signal generator S1 directly (no amplifier) to the FieldFox RF INPUT (50 Ω) using <u>two cables</u> joined by a female-to-female barrel. Place a 10 dB attenuator at the input port of the SA. This will protect the SA from high power levels which we will need when testing the amplifier's compression. So your current set up is: S1 → Cable → F-F barrel → Cable → 10 dB attenuator → SA input port (Port 2). Set S1 to $f_0 = 1.0$ GHz at a low power level of −30 dBm. Set the center frequency to 1.0 GHz, span to 5 MHz, RBW to 30 kHz and leave VBW to `Auto`.

2. Place a marker on the peak (`Marker` → `Peak Search`) and record the measured power P_{ref}. This is your through reference which measures the loss in the cables+attenuator. Remove this loss factor for subsequent DUT measurements. **Turn off the S1 RF power.**

3. Replace the F-F barrel with the RF Amplifier. Before connecting the power supply to the amplifier, check to make sure the DC voltage settings on the power supply are correct. Turn off the power supply, make the DC connections to the amplifier and then turn on the DC supply. Check to make sure the current drawn is according to the amplifier's datasheet. **Double check the system to make sure the RF source power is −30 dBm and the attenuator is connected. Also check that the internal attenuator of the SA (under `Scale/Amptd`) is set to `Auto`.** Then turn on the RF Source. Adjust the scale to see the full signal and the noise.

4. With S1 still at −30 dBm, record the new marker peak power P_{out}. The small-signal gain is $G = P_{\text{out}} - P_{\text{ref}}$ (dB), corrected with the loss factor from earlier. Record G (**Q3.1.1**). Check the value against the datasheet.

3.2 Part B: 1 dB Compression Point

5. Keeping the amplifier and analyzer settings fixed, increase the input power from S1 in 2 dB steps, starting from −30 dBm. **DO NOT MAKE A MISTAKE AND ENTER A HIGH POWER LEVEL. THIS CAN HAPPEN IF YOU MISS THE MINUS SIGN.** At each step, use `Marker` → `Peak Search` and compute the Gain. Take smaller (1 dB) steps as the gain begins to drop. You can turn on the `Signal Track` option which changes the center frequency and the marker frequency to track the signal peak (`Mkr/Tools` → `More` → `Signal Track`).

6. Continue until the measured gain has dropped by 3 dB from the small-signal value. **Do not exceed the maximum input power rating of the amplifier or the analyzer.**

7. Plot P_{out} (dBm) versus P_{in} (dBm), where P_{out} is compensated by the loss of the line. On this curve extrapolate the straight line from the linear region. Determine the input 1 dB compression point $P_{\text{1dB}}^{\text{in}}$ (the input power at which the measured output falls 1 dB below the ideal straight line) and the corresponding output 1 dB compression point $P_{\text{1dB}}^{\text{out}}$. Report both (**Q3.2.1**).

3.3 Part C: Two-Tone Intermodulation of the Test Setup (No Amplifier)

Before measuring the amplifier, we must confirm that the test setup itself — the two sources, the power combiner, and the analyzer front end — is not generating IM3 products that would masquerade as the amplifier's distortion.

8. We now do the two-tone test. Use the external signal generator S1 as one source and the FieldFox internal source + LPF as the other. Set S1 to $f_1 = 1.001$ GHz and the internal source to $f_2 = 0.999$ GHz (a tone spacing of $\Delta f = 2$ MHz). Combine the two outputs in the power combiner and connect the combined signal to the FieldFox RF INPUT (50 Ω, Port 2). Adjust the level of each source so that each tone reads −10 dBm on the SA. **Never let the combined power at the RF INPUT approach +20 dBm, and apply no DC.**

9. The power combiner introduces roughly 6 dB of loss (total loss with cables is around 10 dB) but, more importantly, provides isolation between the two sources. Good source-to-source isolation is essential; otherwise one generator drives the output stage of the other into nonlinearity and produces intermodulation *inside the sources* that would corrupt the measurement. Does terminating the other two ports of the power combiner change the result? What does this tell you about the isolation between the combiner ports (**Q3.3.1**)?

10. Set the center frequency to 1.000 GHz. Reduce the span (e.g. to 10 MHz) and the RBW until the third-order products at $2f_1 - f_2 = 1.003$ GHz and $2f_2 - f_1 = 0.997$ GHz are clearly visible above the noise (use a small enough RBW that they clear the skirts of the IF filter, and turn on trace averaging if needed). Plot the display and record the level of the IM3 products relative to the two tones (**Q3.3.2**).

11. To determine if the IM3 products are generated by the SA itself, place markers on one fundamental and on the adjacent IM3 product. Set `Scale/Amptd` → `RF Attenuation` → `Manual` and step the input attenuation up by 5 dB steps. If the IM3 level drops, the analyzer's own front-end mixer is generating part of it; increase RF attenuation until the IM3 level stops changing, so the front end no longer contributes. If the SA is now not contributing to IM3, where does the residual IM3 come from (**Q3.3.3**)?

12. Record the residual (setup) IM3 level relative to the tones. This is your distortion floor: any amplifier IM3 you measure later must sit well above this level to be accurate. Turn off the sources.

13. Record the power coming out of the power combiner for each tone to note the loss through the combiner for each tone. You will need this number for the next section.

3.4 Part D: Amplifier Third-Order Intermodulation and IIP3

14. Insert the amplifier into the two-tone path: connect the combiner output to the amplifier input, and the amplifier output through the **10 dB attenuator** to the RF INPUT (Port 2), as in Parts A–B. Confirm the DC bias current matches the datasheet. Set the power *per tone* at the amplifier input well below the 1 dB compression point found in Part B, e.g. $P_{\text{in}} = P_{1\text{dB}}^{\text{in}} - 15$ dB.

15. Verify that the measurement reflects the amplifier and not the setup by turning on both sources and observing the IM3 tones. Explain why this check is necessary (**Q3.4.1**).

16. **Sweep the input power and record both the fundamental and the IM3 output.** Keeping the two tones equal in amplitude at every step, increase the per-tone input power from the low starting level of -15 dBm in 2 dB steps (take smaller, 1 dB steps once the fundamentals begin to compress). **DO NOT MAKE A MISTAKE AND ENTER A HIGH POWER LEVEL RIGHT AWAY. THIS CAN HAPPEN IF YOU MISS THE MINUS SIGN**; raise both generators together so that the two tones stay balanced. At each step, use the markers to read the output power of one fundamental tone, P_{fund}, and of the adjacent IM3 product, P_{IM3} (both corrected for the loss factor), together with the per-tone input power P_{in} at the amplifier input (corrected for the combiner loss).

17. Continue stepping the power up until the fundamental output has just begun to compress (its gain has dropped by roughly 1 dB) or the IM3 output visibly departs from its steep straight-line growth, then stop. **Do not exceed the maximum input power rating of the amplifier or the analyzer, and do not drive the amplifier into hard compression. Adjust the scale on the SA if needed.**

18. Plot P_{fund} (dBm) and P_{IM3} (dBm) versus the per-tone input power P_{in} (dBm) on the same axes. In the small-signal (linear) region — below the onset of compression — the fundamental data should follow a straight line of slope 1 dB/dB and the IM3 data a straight line of slope 3 dB/dB. Fit a straight line to the linear-region points of each trace and *extrapolate both lines* until they intersect; the intersection is the third-order intercept point. Read the output intercept OIP3 off the vertical axis and the input intercept IIP3 off the horizontal axis at the crossing point, and report both. As a consistency check, confirm that a single mid-range point satisfies $\text{IIP3} = P_{\text{in}} + \Delta P/2$ and $\text{OIP3} = \text{IIP3} + G$, with G the gain from Part A (**Q3.4.2**).

19. From your plot, over what range of input power do the fundamental and IM3 traces actually follow the ideal 1 dB/dB and 3 dB/dB slopes? Explain what happens to each trace as the amplifier approaches compression, and why only the linear-region points may be used in the extrapolation (**Q3.4.3**).

20. Compare your measured IIP3 with the 1 dB compression point from Part B. By how many dB does IIP3 exceed $P_{1\text{dB}}^{\text{in}}$? Compare this with the commonly cited rule of thumb ($\sim$9.6 dB for an ideal third-order-limited device) and comment on any difference (**Q3.4.4**).

Reminder: Take turns in doing the experiments.

4 External Millimeter-Wave Harmonic Mixing

The FieldFox measures signals directly only up to 26.5 GHz. To reach millimeter-wave frequencies the signal has to be downconverted *before* it arrives at the analyzer, which is the job of an external harmonic mixer. A low-frequency LO is supplied to the mixer on coax, the mixer generates and uses the Nth harmonic of that LO, and the product that falls inside the analyzer's own frequency range is returned to it:

$$f_{IF} = \left|f_{RF} - N f_{LO}\right|. \tag{1}$$

The downside is that Eq. (1) has many solutions. For a given LO, *two* RF frequencies convert to the same IF, $N f_{LO} - f_{IF}$ and $N f_{LO} + f_{IF}$ — the two sidebands — and such a pair exists for every harmonic N present in the mixer drive. The mixer is **unpreselected**: there is no tunable filter ahead of it to select one response and reject the others. All of them therefore reach the analyzer, and a single input tone can paint several lines on the screen, only one of which is at the correct frequency. Here we will see how one can tell the real signal from the image signals.

The VDI PSAX Module

The module used here is a Virginia Diodes Portable Spectrum Analyzer Extension (PSAX). It bolts directly onto the two RF ports of the FieldFox: Port 1 supplies the LO and Port 2 receives the IF. Its key specifications are collected in Table 2; read the datasheet and the operation manual before connecting anything.

Parameter	**Specification**
RF frequency band	60–90 GHz (WR-12, UG-387/U-M flange)
LO harmonic factor N	6
LO input	0 dBm ± 3 dB typical; **damage at** +8 **dBm** (2.92 mm)
IF output	100 MHz to 9 GHz (2.92 mm)
RF input limits	−10 dBm ($P_{1\text{dB}}$); **damage at** +17 **dBm**
SSB conversion loss	−10 dB typical — negative loss is *gain*
Noise figure	12 dB typical
DC bias	+5.5 to +7.5 V, ≈300 mA (LEMO 00)

Table 2: VDI WR-12 PSAX module, key specifications.

Two features matter for what follows. First, the LO is multiplied by ×6 *inside* the module and that multiplied output drives the mixer, so $N = 6$ is strongly favoured over the neighbouring harmonics rather than being merely one option among many. Second, the module contains an IF amplifier, so it has conversion *gain*: the datasheet quotes a typical SSB conversion loss of −10 dB, i.e. about **10 dB of conversion gain**. The FieldFox does not know this so it must be compensated for externally.

Frequency Planning: This Is Not the Fixed-IF Architecture

In the Signal Identification method discussed in the lecture the IF was *fixed*: the LO tunes, the IF filter sits at one frequency, and the image is whatever lands $2f_{IF}$ away from the wanted signal. This is a widely used architecture on the big box spectrum analyzers. However, here the IF port of the PSAX is connected to Port 2 of the FieldFox, which is itself a complete 30 kHz–26.5 GHz spectrum analyzer, while the LO comes from Port 1 and is tunable over the same range. *Both* the LO and the IF can therefore span a broad range, and the instrument is free to choose whichever pair (f_{LO}, f_{IF}) suits the sweep it has been asked to make. The configuration file makes this explicit: it specifies an IF *range* (typically 0.3–9 GHz) and an LO range, not a single IF. You will see the consequence directly in Step 4, where the analyzer does not necessarily keep the same IF when you change the center frequency.

What does *not* change is the problem of images and spurs. Equation (1) still has many solutions and an unpreselected mixer still responds to all of them. It is worth writing down where they land. If the analyzer is configured for harmonic N and is using an intermediate frequency f_{IF}, then a real signal at f_{RF} that converts through harmonic n is displayed at

$$f_{\text{disp}} = \frac{N}{n}\big(f_{RF} \mp f_{IF}\big) + f_{IF}. \tag{2}$$

For $n = N$ and the correct sideband this collapses to $f_{\text{disp}} = f_{RF}$: the IF cancels and the true response is displayed at the right frequency whatever LO/IF pair the instrument has chosen. Every other response keeps a term in f_{IF}. In particular the opposite-sideband image of the wanted signal appears at

$$f_{\text{image}} = f_{RF} \pm 2f_{IF}, \tag{3}$$

which gives a direct way to read the IF off the screen.

Safety and equipment limits for this section. The PSAX and PSGX modules are expensive and easily damaged. Never exceed +8 dBm of LO drive or +17 dBm at the RF flange, set each module's DC bias to the voltage specified for it below, and never connect or disconnect DC while a module is powered. Handle waveguide flanges by their bodies, inspect them before mating, and torque the waveguide screws using the provided torque wrench. **For this experiment you will need to remove the port protectors, so please ask the instructor to help you with this.**

Experiments

In these experiments a **PSGX** source module transmits a CW tone at **72 GHz** and the PSAX receives it. Please ask the instructor to give you the equipment needed for these experiments. You will have to take your FF and go to the dedicated lab station for this experiment.

1. **Connect and configure the PSGX.** Connect the RF input of the PSGX to Port 1 of the FieldFox using an SMA cable, and connect the DC cable from the DC output on the left side of the FieldFox to the DC input on the PSGX. `Preset` the instrument, then set the mode and the DC bias:

   `Mode` → `Pulse Generator`
   `System` → `System Configuration` → `Voltage Source`
   → `Voltage` → **9 V** → `Source ON`

   Then set the drive frequency and amplitude, and turn on the RF:

   `Freq/Dist` → 12 GHz
   `Amptd` → −5 dBm
   `Measure` → `RF ON`

The PSGX multiplies this drive by six, so it is now radiating at 72 GHz.

2. **Connect and configure the PSAX.** Ask the instructor to remove the port protectors from your FieldFox, then mount the PSAX directly onto the two RF ports — Port 1 to `LO` and Port 2 to `IF` — and connect the DC cable from the DC output on the left side of the FieldFox to the DC input on the PSAX. `Preset` the instrument, then set the mode and the DC bias:

   `Mode` → `SA`
   `System` → `System Configuration` → `Voltage Source`
   → `Voltage` → **5.5 V** → `Source ON`

   Now load the head file for the module and turn on Frequency Extension mode:

   `System` → `Utilities` → `Freq Extnd`
   → `Select Freq Extnd Head` → `60-90 GHz` → `Recall File`
   → `Freq Extnd ON`
   `Freq/Dist` → `Center` → 72 GHz

   Check the `LO Power` setting and make sure it is −10 dBm. This is below the 0 dBm drive the module specifies, so the PSAX is deliberately under-driven here; that is fine for these measurements, though the conversion gain will be slightly lower than the datasheet value. Point the two antennas towards each other; you should see a strong signal at 72 GHz. You are now ready with your mm-wave link!

3. **Measure the 72 GHz signal.** Remember that the PSAX contains an internal IF amplifier, so the module has roughly 10 dB of conversion *gain* at 72 GHz rather than a loss, and the FieldFox does not correct for it. Record the displayed power and give the corresponding power at the PSAX waveguide flange (**Q4.1.1**).

4. **Find the image, and watch it move.** Keeping the transmitter fixed at 72 GHz, change the analyzer's center frequency and look for a second response of nearly equal amplitude. Note that the frequency of this image response *changes* when the center frequency of the instrument is changed, while the true 72 GHz response does not. Explain why, using Eqs. (2) and (3) (**Q4.1.2**).

   (a) Set the center frequency to 77 GHz. Record the image frequency, and from it estimate the IF and the LO frequency the instrument is applying (**Q4.1.3**).

   (b) Repeat with the center frequency set to 79 GHz (**Q4.1.4**).

   (c) Keep changing the center frequency and look for any further image or spur responses (which would be weaker than the main image). Record the frequency and amplitude of one such spur, and use Eq. (2) to guess which harmonic n produced it (**Q4.1.5**).

5. **Plan the conversion yourself:** $N = 6$. Turn Frequency Extension mode `OFF` and return to ordinary SA mode. The FieldFox now displays the IF directly and you supply the LO yourself from Port 1 (`Measure` → `Source`). If the RF is at 72 GHz and you want an IF of 1 GHz with $N = 6$, what should the LO frequency be? Calculate it from Eq. (1), apply it from the FieldFox internal source with a power level of −10 dBm, set the analyzer center frequency to 1 GHz with a narrow span, and record the signal power (**Q4.1.6**).

6. **The image, by hand.** Now set the LO so that the same 72 GHz signal reaches the 1 GHz IF through the *image* sideband, still at $N = 6$. Calculate the new LO frequency, apply it, and record the power. Is it the same as in Step 5? Explain what you find (**Q4.1.7**).

7. **A different harmonic.** Set the LO so that the 72 GHz signal is received on the *lower* sideband of $N = 7$ at an IF of 1 GHz, i.e. $7f_{LO} - f_{IF} = f_{RF}$. Calculate the LO frequency, apply it, and record the power (**Q4.1.8**). How far below the $N = 6$ response does it sit, and what does that tell you about the $\times 6$ multiplier chain inside the module?

When you are finished, turn off the PSGX and PSAX LO/RF sources first, then the DC bias to both modules, disconnect the instruments and ask the instructor to replace the port protectors.

Appendix

A Percent of Harmonic Distortion

The total percent of harmonic distortion of a signal is also measured frequently. For this measurement, the amplitude of each harmonic must be measured in linear units (for example, volts) instead of dBc. To display amplitude units in volts, press `Scale/Amptd` → `Y Axis Units` → `Volts`.

The amplitude values of these signals are used in the equation below to compute the total harmonic distortion:

$$\text{Percent of distortion} = \frac{\sqrt{(A_2)^2 + (A_3)^2 + (A_4)^2 + \cdots + (A_n)^2}}{A_1} \times 100 \tag{4}$$

where:

$A_1 =$ the amplitude of the fundamental frequency, in volts

$A_2 =$ the amplitude of the second harmonic, in volts

$A_3 =$ the amplitude of the third harmonic, in volts

$A_4 =$ the amplitude of the fourth harmonic, in volts

$A_n =$ the amplitude of the nth harmonic, in volts

If the signal amplitudes are measured carefully, as in Procedure B (Section 2.2), this procedure measures percent-of-harmonic distortion very accurately (**Q2.2.3**).

A.1 Converting dBc Readings to Percent Distortion

Method 1 of Procedure A and Procedure B read each harmonic relative to the fundamental in dBc. Let $\Delta_n = P_n - P_1$ be the level of the nth harmonic relative to the fundamental (in dBc, a negative number). Because power ratios in dB are $20\log_{10}$ of voltage ratios, $A_n/A_1 = 10^{\Delta_n/20}$, and the percent of distortion due to the nth harmonic alone is

$$\text{Percent of distortion}_n = 100 \times 10^{\Delta_n/20}. \tag{5}$$

Substituting $A_n/A_1 = 10^{\Delta_n/20}$ into Eq. (4) gives the total directly from the dBc readings:

$$\text{Percent of distortion} = 100\sqrt{\sum_{n=2}^{N} 10^{\Delta_n/10}}. \tag{6}$$

For example, a single harmonic at −40 dBc corresponds to 1% distortion.