

## Carrier Sleuthing, part#3

### **Spectrum Analyzer Software**

Be sure to read the other parts of this series. [Part#1](#), [Part#2](#), [Part#3](#)

Online references to carrier sleuthing usually assume that the signal analysis is done at some time post-reception, via a recording, and almost always using a modern SDR receiver (the SDR's I-Q data recording). It is also usually assumed that we are looking at a waterfall histogram for carrier peaking in order to copy actual faded-up voice to enable audio verification of a station.

That's not what our thrust is here.

The thrust of this series is real time weak carrier sleuthing, that is, DXing weak carriers as they are actually being transmitted. Using schedules, sunrise-sunset maps, propagation maps, offset charts and detective work we will deduce which station's carrier we are seeing. We may often watch the same frequency over several days or even weeks to see what changes occur, what signals come and go, in order to refine our analysis.

We complete this process with a marvelous range of spectrum analyzer software available to us, and it's all free. These are some of the more popular and useful ones:

- SBSpectrum
- Argo
- Spectran
- SpectraVue
- Spectrum Lab
- Lopora

All of these will require a certain amount of initial "setup" to get them functional and show the resultant graphic waterfall we want. In your hands you hold professional-grade spectrum analysis capability, and all for free.

I actually use two and sometimes three of these at a time. I use the simple SBSpectrum as a rough guide to locate the carrier offset, then knowing the carrier offset, I use Argo or SpectraVue or Spectrum Lab to home in on the carrier. A good choice for the beginner is SBSpectrum and Argo, or SBSpectrum and Spectran. They are simple to set up.

Our purpose is to analyze the audio stream coming from our radio, whether that radio is a physical radio or an SDR radio. In some fashion we will plug that audio coming out of our radio directly into our computer's audio input.

I know it's painful, but it's helpful to know the basics about FFT - Fast Fourier Transform. I'll try to keep it brief.

## **FFT Basics**

Simply stated, FFT is short for Fast Fourier Transform. It's the mathematical algorithm that enables spectrum analyzer software to convert streamed analog audio signals into numerical representations of its frequency components. These numerics are stored in an array of "frequency bins". They are complex numbers representing magnitude and phase at each frequency bin. The term "FFT" has also come to mean the set of "frequency bins". "FFT size" is number of points or "bins" our FFT holds.

Spectrum analyzer software samples incoming audio at a prescribed rate (typically 48000 samples per second) and applies the FFT algorithm to blocks of this audio data. Numerical representations of these blocks are then distributed to the frequency bins. Each FFT bin represents a specific frequency range, not a single frequency

point. For example, if our FFT is sized at 16384 points (or bins) and we sample at 48000, it provides frequency resolution of about 2.93 Hz per bin across a 0-24000 Hz spectrum ( $48000/16384$ ). Note that the bandwidth of spectrum able to be analyzed is always half of the sample rate. At 48000 samples input, half is 24000. This is called the "Nyquist" frequency, in this case, 24 kHz.

If we're looking for carriers, do we need to see 24 kHz of spectrum bandwidth? No, not really. We're almost always checking a specific frequency for carrier activity. 3 kHz or even 6 kHz is adequate. Our physical radio will probably have a SSB filter in this range and we will tune the receiver slightly below the carrier so our carrier is somewhere near the center of the bandpass. If you're using an SDR, choose a 3-6 kHz bandpass filter. Our spectrum analyzer software can then look at that small slice of spectrum presented to it by the radio.

Advanced spectrum analyzers like most of the above allow user-chosen FFT sizes (1024 to 65536 points or more) allowing us to trade between frequency resolution and time resolution. Longer FFTs provide finer frequency discrimination but at slower update rates. The software often includes averaging capabilities, accumulating multiple FFT results to reduce noise and reveal weak signals buried in the noise floor. For carrier sleuthing, an FFT size of 65536 is a good choice and a good starting point.

To improve spectral purity and increase frequency resolution the software can apply windowing functions to each data block before processing by the FFT algorithm. These are the familiar Hamming, Hann, Blackman windows, etc. Windows are typically overlapped, processing new FFT blocks every 25-50% of the window length rather than waiting for complete non-overlapping blocks. This overlap increases the effective update rate and smooths the spectral display.

When the input signal doesn't contain an integer number of cycles within the FFT window, energy spreads across multiple bins - this is spectral leakage. The rectangular window (no windowing) acts as a sinc function in the frequency domain, causing the familiar "picket fence" effect where true signal frequencies between bin centers are attenuated.

Windowing functions like Hamming or Blackman multiply the time-domain data before the FFT, trading main lobe width for reduced side lobe levels. This changes how signal energy distributes across bins but reduces leakage artifacts.

Finally, frequency bin data is converted to density values based on spectral power and is often scaled to decibels, or dB. The software maintains a scrolling waterfall display by stacking successive FFT results vertically, with frequency on the horizontal axis and time progressing downward, using color palette mapping to represent signal strength.

In practice, the software maintains multiple processing threads:

1. Input Thread: Continuously fills circular buffers with incoming samples
2. FFT Thread: Applies windowing, performs FFT calculations, and computes bin magnitudes
3. Display Thread: Converts bin data to pixel values and updates the visual display

The software may also support features like offset correction, automatic gain control, hum filters, notch filtering of interference, time stamping, auto-screen capturing, outputting sample data to a file, and more. There are usually a lot of options in settings to experiment with once you get the basics down.

## **Common Settings: Set the Input Device**

Common to all and what needs to be set up first is the sound card configuration. All modern computers have a "sound card", or sound input/output ability of some sort. A sound card (or "chip" these days) performs two primary operations: on output, it converts digital audio data from your computer into analog signals that can drive headphones or speakers. On input, it converts analog audio signals from microphones or other sources into digital audio data the computer can process.

We must always tell our spectrum analyzer software from where its input is coming. It will be your microphone or your virtual cable. Older computers may still have a line input. Look in the settings area of the software to set your input device.

## **Common Settings: Set the Sample Rate**

The sample rate we choose and the size of our FFT is important in carrier sleuthing in order to get the best results. Please follow along. We must always tell our spectrum analyzer software at what sample rate it should sample the incoming audio stream. Look for this in the settings area of the software.

An end sample rate of about 6000 samples per second I feel is optimal for our carrier sleuthing purposes, and it produces good results. We can get that in a couple of ways. Most sound cards default to 48000 samples per second (DVD, or professional audio), way too high for our needs. It's best to leave them at their default, in part because other programs on your computer may prefer this setting, like your music player. We will use the "decimation" function in our spectrum analyzer to divide this down to the sample rate we want.

The decimation value we want is 8, which will divide 48000 by 8, resulting in 6000 samples per second. Simply put, 6000 samples per

second gives us a 3000 Hz bandwidth (remember, it's always half the sample rate). This fundamental limit comes from the mathematics of signal processing and cannot be overcome - it's a law of nature for sampled systems. 3000 Hz is a nice slice of spectrum we can make available to view in our spectrum analyzer.

Decimation can usually be set on the incoming audio sample rate, but is often also found as an FFT setting. If both are offered, use one or the other, not both.

### **How Decimation Works (technical, skip if you like)**

Decimation is a two-step process that reduces the sample rate while preventing aliasing:

#### **Step 1: Anti-Aliasing Filtering**

Before reducing the sample rate, the signal is low-pass filtered to remove frequency components above half the new sample rate (the new Nyquist frequency). For decimation by 8 from 48 kHz to 6 kHz, you'd filter out everything above 3 kHz to prevent aliasing.

#### **Step 2: Downsampling**

After filtering, you simply keep every 8th sample and discard the rest. This is the actual sample rate reduction. Without the anti-aliasing filter, high-frequency components would fold back into the lower frequency range as aliases, corrupting the signal. The filter ensures that only frequencies that can be properly represented at the new sample rate remain in the signal.

### **Practical Implementation**

Modern decimation is often implemented using efficient polyphase filters that combine the filtering and downsampling operations, or through cascaded stages (like decimating by 2 three times instead

of 8 once) to reduce computational complexity while maintaining good anti-aliasing performance.

### **Common Settings: Set the Center Frequency**

In USB mode again, if we tune our receiver to 1 kHz below the expected carrier's frequency the carrier should show up at 1000 Hz on our spectrum and waterfall display if they are transmitting. Set the center frequency accordingly to place the frequency of interest in the center of the window.

### **Common Settings: Set the Span**

Most spectrum analyzers have a separate span setting. The "span" is the amount of spectrum we wish to view in the spectrum and waterfall display, either side of the center frequency. To start with, set it wide, between 200-1000 Hz. Stations, particularly shortwave, can sometimes be quite a bit off frequency. Setting to 200-1000 Hz will enable us to see a greater range either side of the frequency of interest. Once the carrier is found, we can recenter the carrier and then reduce the span down to 100 Hz or lower (even down to almost 0). At lower spans you may see doppler and auroral shifts, anomalies in the transmitted carrier, or other carriers only a few cycles away, etc. Exciting stuff!

## **The Programs**

Now let's dive deeper into the actual programs. I'll give you detailed set up information for each. I will assume you know how to install software on Windows.

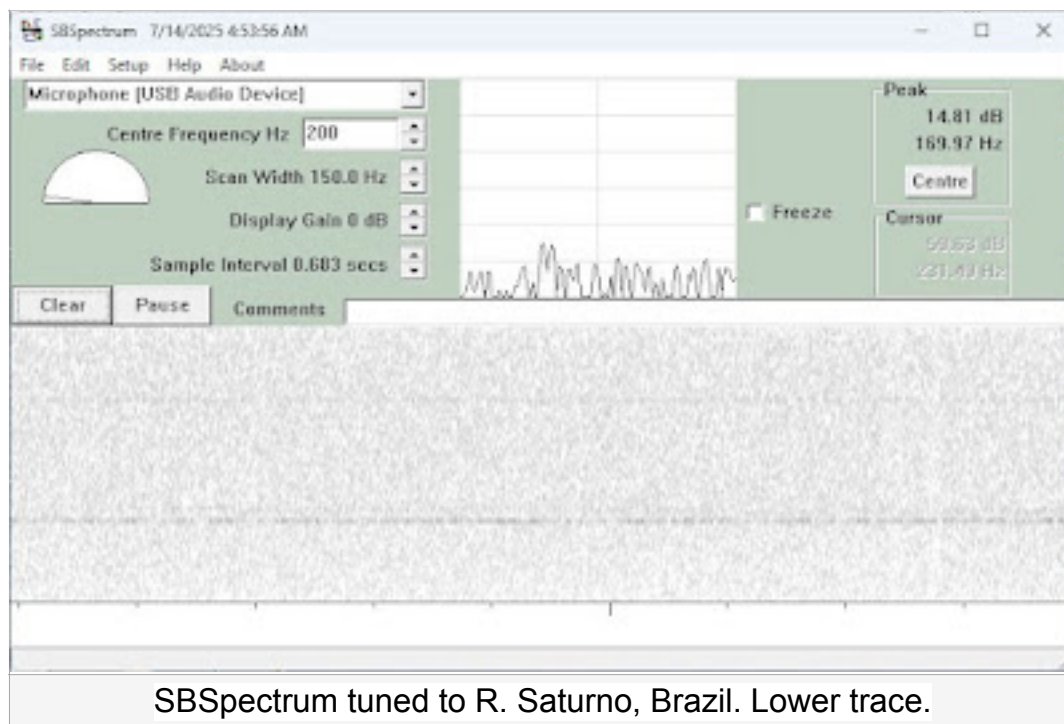
The following thumbnail images of each spectrum analyzer program in action show reception of pirate Radio Saturno 6150 kHz, Belo Horizonte, Brazil, on July 14, 2025. Signals were captured at local sunrise time here in western New York, which coincided very closely at both locations. The station was approximately 30 cycles off (low) in frequency. The receiver, an SDRPlay RSP1B in USB mode, was

tuned 200 Hz below 6150 kHz, centering Saturno's carrier in the RSP1B's 400 Hz bandpass filter. The antenna is a one turn passive loop.

Click any of these thumbnails to get the bigger picture.

## SBSpectrum

SBSpectrum, by Peter Martinez, G3PLX, is perhaps the handiest and simplest software to see carriers play out on a strip chart-like display. It was written many years ago to look at carrier doppler shifts, called "dopplergrams", in the HF spectrum. It is Windows only and runs on any Windows installation at least back to Windows XP. No install is necessary, simply unzip the downloaded file and click on the executable. It is bare bones, but works flawlessly.



I'm using SBSpectrum 1.31. A recent copy of the SBSpectrum executable is also available here, along with a Help file:

[SBSpectrum 1.31](#) (via creator's Dropbox link)



You can download the latest SBSpectrum version from the [google.io Dopplergram group](#), but you need to be a member. Look in the "Files" section. The group is worth joining as there are other resources, and members are willing to provide guidance and help interpret what you see.

Another nice [Help file can be found here](#).

I'll give a fairly detailed setup procedure as SBSpectrum has simple controls. The other spectrum analyzer software packages can be much more detailed and some of their controls and settings you will have to dig for in their menu systems.

In USB mode on your receiver, tune to 1 kHz below the carrier of interest.

In the dropdown box, tell SBSpectrum where its audio input is coming from (the microphone or the virtual cable output).

In Settings, set the sample rate. Set Nominal sample rate to 6000 and Actual sample rate to 6000.

Back to SBSpectrum, set the center frequency to 1000.00.

Set the initial Scan Width to 1200 Hz.

Make sure the little meter window at the left shows evidence of signal coming in. If not, increase the input gain some (use your radio or SDR volume control) so the meter needle wiggles a little within the lower 1/3 part of its scale. You should also see a wiggly waveform in the top center window. Align this wave form with the bottom of the window by ticking the Display Gain up or down. Note that the little meter is also a handy little gadget enabling you to set the proper input level for all other spectrum analyzer software.

Look for a carrier on the main display, a darker line on the waterfall's horizontal scroll. If strong enough, its peak frequency will register

unwaveringly in the "Peak" information box. You can click the "Centre" button to center the carrier. You can also click the signal peak in the spectrum display to center its carrier, or even click the carrier itself in the waterfall display.

The Sample Interval can be chosen, but should automatically set to the optimum value after setting the Scan Width.

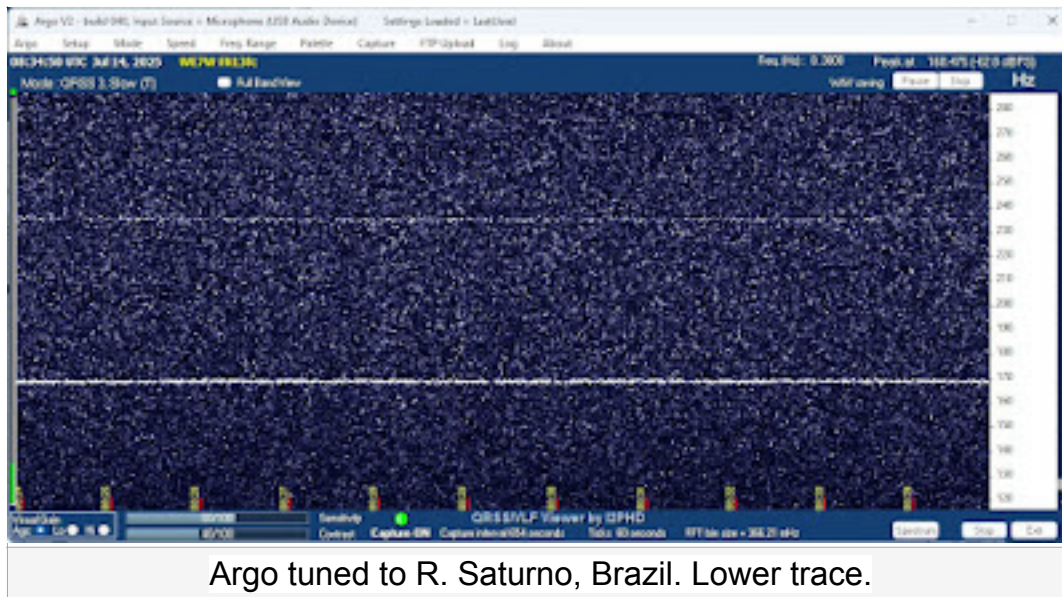
Since we tuned our receiver at 1 kHz below the carrier frequency, the carrier's offset referenced to 1000 Hz is its frequency offset. Once we center the carrier of interest we can use this center peak value to locate the carrier in one of the other spectrum analyzer software apps.

With success, your display should have a very light background and a dark or even smudgy carrier line scrolling across. Adjust Display Gain for best looking display.

My usual carrier sleuthing habit is to locate the carrier of interest in SBSpectrum, then use that location offset in one of the more comprehensive analyzers.

## **Argo**

I use the Argo spectrum analyzer program (V2, build 140) for carrier sleuthing. It is written by Alberto, I2PHD. It works wonderfully for carrier sleuthing using its QRSS 3 mode. It is very easy to set up and runs on almost any version of Windows. These instructions are for V2 build 140.



[The Argo program can be found here.](#)

Argo has evolved over the years. It was written for ham operators to view QRSS signals. QRS (one 'S') is the old telegrapher's "Q" code for "send slower". QRSS corrupts that to mean "send extremely, extremely slowly". By that they mean 3 second long dots or slower. It is usually sent as FSK, or frequency shift keying, dots and dashes shifted by 5 Hz. The slow rate greatly reduces the bandwidth of the carrier and increases its readability. The claim for QRSS is that a carrier signal can be received at -26 dB below the noise. Very low transmitter power is usually used, sometimes micro watts.

Sending QRSS signals is a ham radio sub-hobby. To transmit them you'll need a ham radio license. But of course we might use the excellent Argo receiver software for our carrier sleuthing purposes - no license required. More on QRSS and its technical application can be found at these links:

[The New Age of QRSS](#)

[QRSS and You](#)

Let's set up Argo for carrier sleuthing.

In USB mode on your receiver, tune to 1 kHz below the carrier of interest.

It's not possible to set Argo's audio or FFT sample rate, other than choosing different reception modes. Argo uses an internal sample rate of 6000 samples per second.

Under Select Sound Card in Setup, choose either microphone or the virtual cable.

Set the input sound level (use your radio or SDR volume control). You will also see a varying vertical green bar along the left side of Argo's window. This is the input gain. It should be between  $1/8$  and  $1/2$  of the total height. 20%, or  $1/5$  is a good height. You can also use the SBSpectrum meter to see the level.

Uncheck FullBandView. This will give you a horizontal waterfall.

Set Visual Gain to AGC or High.

Set Sensitivity to 60.

Set Contrast to 40.

Under Mode, set to QRSS 3.

Under Speed, set to Slow.

Under Palette, set to Dark.

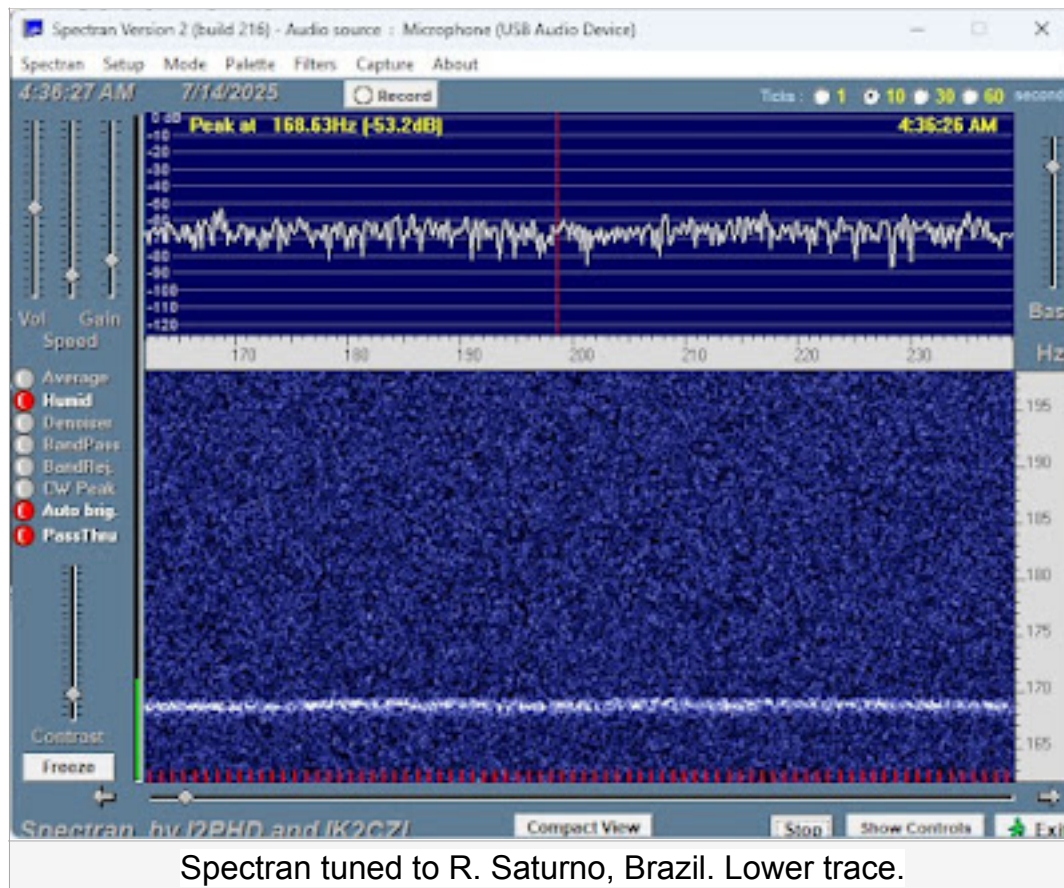
Click the Start button and the waterfall scroll will commence. The vertical offset scale on the right can be scrolled with the mouse. Find the carrier and center it.

The new version of Argo also has a nice spectrum display. Access it by clicking the Spectrum button.

Check out Argo's other settings for more features like time stamping, screen capturing, etc.

## Spectran

Spectran also runs on Windows and is relatively easy to set up. Spectran is also written by Alberto, I2PHD. I'm using Version 2 build 216.



[The Spectran program can be found here.](#)

In USB mode on your receiver, tune to 1 kHz below the carrier of interest.

Under Select Sound Card in Setup, choose either microphone or the virtual cable.

Set Volume slider at center.

Set Speed slider at the 3rd tick from the bottom.

Set Gain slider at the 3rd tick from the bottom.

Set Contrast slider at the 3rd tick from the bottom.

Set Base slider to the 3rd tick from the top.

Set the input sound level (use your radio or SDR volume control).  
You can use the SBSpectrum meter to see the level.

Click Show Controls to bring up a little window with more settings.

Check 8000 samples per second. 6000 is not available with Spectran.

Set Avg Factor to 1.

Check Moving Average.

Set Scrolling to Hor (Horizontal).

Check USB.

Set Resolution to 0.12 Hz. That will force our FFT size to 65536.

Hide the Show Controls window.

Under Palette, set Standard.

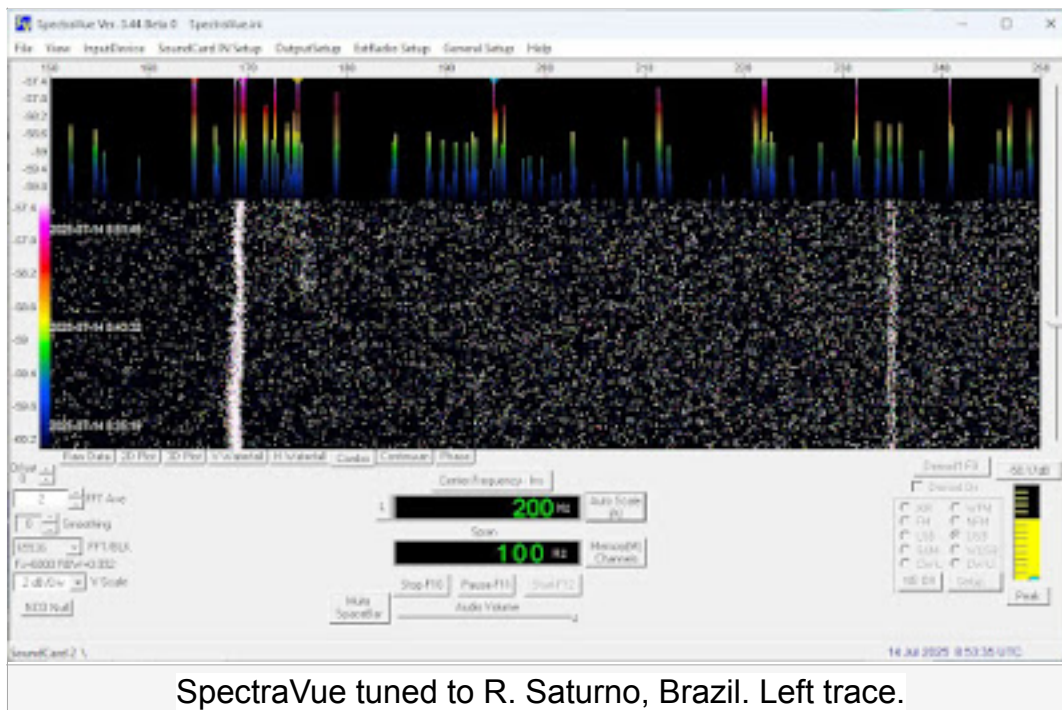
Click the Start button and the waterfall scroll will commence. The horizontal frequency scale above the waterfall can be scrolled with the mouse or the lower scroll bar. Find the carrier and center it.

Be sure to check out other settings for more features.



## SpectraVue

SpectraVue was originally written for the first RF Space SDR receivers, the SDR-IQ and the SDR-14. It is an excellent spectrum analyzer in its own right and is easy to set up. It's Windows only again. I'm using version 344.



[The SpectraVue program can be found here.](#)

In USB mode on your receiver, tune to 1 kHz below the carrier of interest.

Under Input Device, check SoundCard.

Under "SoundCard In Setup":

Choose either microphone or the virtual cable.

Set the Sample Rate to 6000 samples per second.

Set the BW Limit to 3000 Hz.

Keep Center Freq at 0.

Under General Setup:

Set FFT Window Type to Hamming.

Set Display Units to Hz.

Check Color 2D Graph.

Set FFT Overlap to 1.

Set Skip N Updates to 0.

Set Waterfall Rate to 40.

Check Display Peak Markers.

Set Mouse Click resolution to 1 Hz.

Check Time Stamp Display, 0 Sec (0=Auto).

SpectraVue has the ability to demodulate I/P data coming from the RF Space SDRs. We can ignore this entire demodulation function as we will only be using audio input.

On the main window:

Click the Combo tab.

Set FFT Ave to 2.

Set Smoothing to 0.

Set FFT BLK to 65536.

Set V Scale to 2 dB/Div.



Under Center Frequency, set to 1000.

Under Span, set to 200.

Set the input sound level (use your radio or SDR volume control). You can use the SBSpectrum meter to see the level. The audio spectral waveform will be at the top of the window. A properly adjusted input gain will have the waveform in the lower 1/3 of the scale.

Click Start to start the waterfall analysis.

Clicking Auto Scale will auto-adjust the waterfall gain. The slider on the right can be moved up or down for fine gain adjustment. Adjust so the audio spectral waveform is in the lower 1/3 of the scale.

Find the carrier. Center the carrier by resetting the Center Frequency. If you don't see the carrier, widen out the span some, to 600-1000 and check again.

Be sure to check out other settings for more features.

## **Spectrum Lab**

Spectrum Lab is the Big Daddy of them all, and the hardest one to learn. It will take you time and dedication to become fully aware of all its possibilities. If you can get by the steep learning curve on this one, you will be rewarded with amazing capabilities in signal viewing.



Set Input Device / Stream / Driver to either microphone or the virtual cable.

Keep bits/sample at 16.

Keep input chns at 1.

Under the FFT tab:

Set Decimate input by (divisor) to 1 (no decimation).

Set FFT input size to 65536.

Set FFT window function to Hamming or Hann.

Check use anti-alias filter for decimation.

Check same FFT params for all analyser channels.

Ensure Type is Real FFT.

Ensure Source is correct for your setup.

Set internal average to 0.

Set smoothing (#bins) to 0.

Under the Spectrum (1) tab:

Check optimum waterfall average. Leave others unchecked.

Set Waterfall Scroll Interval to 2000 ms (milliseconds).

Uncheck automatic.

Check smooth scroll.

Check Amplitude grid.

Check one pixel per FFT bin. Leave others unchecked.

Set Spectrum graph area (px) to 100.

Check Waterfall time grid enabled.

Uncheck Interval automatic.

Check sec (for seconds). Enter 60 under Interval.

Choose your style and labels for the time stamps.

Under the Spectrum (2) tab:

Set Amplitude Range to -90 and -50 dB.

Set Offset to 0.0.

Set VisualAGC to fast.

Set Ref to -64 dB. This is your AGC threshold level.

Check grid in graph (under Options for the Frequency Axis).

Set Frequency scale style to Ruler.

Set Radio Freq Offset to 0.

Next, we'll tackle the settings on the main window.

The three text boxes are changed simply by entering a new value and pressing Enter.

vfo text box - irrelevant as this setting is used for receiver control.

fc text box (frequency center), set to 1000.

sp text box (span), set to 200.

The key to exceptional signal viewing in Spectrum Lab is getting the palette and brightness and contrast settings dead on. Much depends on the proper palette choice, the AGC versus Range setting, and the width and left-right orientation of the palette which is controlled by the Contrast and Brightness and sliders.

Start with the Brightness and Contrast controls centered.

Set the Palette.

Click in the colored palette display to see a list of possible palettes. The palette I like best and which gives the best display (subjective of course) is "openlab\_electric". It has a range of spectral colors from black to red. Select "openlab\_electric".

Set the Contrast.

I first use the Contrast slider to reduce the palette's black to red transition so it's fairly narrow, then I position this transition using the Brightness slider. If we have our controls set properly, any weak signal will fall into this range and immediately pop out of the black background.

Set the Contrast almost all the way to the right at about the 80% point. This will narrow up our color transition nicely. Note that our dB gauge under the palette thumbnail will read -90 to -50. This is due to our audio input Range limits in Settings.

Set the Brightness.

Adjust the Brightness slider to center the black to red transition between the -70 and -60 dB points, with the red boundary over the AGC threshold value, -64 dB.

During operation, fine tune the brightness control for best viewing. Use Contrast to finely adjust the sharpness of the display.

Set the input sound level (use your radio or SDR volume control). You can use the SBSpectrum meter to see the level. The audio spectral waveform will be at the top of the window. A properly adjusted input gain will have the waveform in the lower 1/3 of the scale.

Under Start/Stop, click Start Sound Thread to start the waterfall analysis.

Find the carrier. Center the carrier by resetting the Center Frequency (fc). If you don't see the carrier, widen out the span some (sp), to 600-1000 and check again.

Be sure to check out other settings for more features.

## **Lopora**

Lopora is special because it runs in the Python language and thus may be run on any system which has Python installed (including Linux and Mac). It is not full-featured, however. Like Argo, it was written for hams to decode QRSS signals. It's simple to set up and does a good job. I'll leave it at that for this one and let you experiment.

[Lopora can be found here.](#)

Some sample settings for Lopora:

Compress 0

Contrast 6

Brightness 1

Sample rate 5512

FFT window 5 (Nuttall)

FFT size (32768, 5=Nutall) or (16384, 1=Cosine)

Windows are 0-6, lower values require lower sized FFT.

## **Carrier Sleuthing Tips**

### **Receiver Bandwidth**

I've been carrier sleuthing at a much reduced bandwidth lately to enhance the signal-to-noise ratio and improve signal sensitivity. Lately I've chosen a receiver bandpass of 1000 Hz. With the RSP1B SDR I can create a 1000 Hz bandpass filter, and after tuning the receiver 500 Hz lower than the carrier of interest, the carrier then falls right in the middle of the filter bandpass. This, in USB mode of course. Note that many SDRs have a low-cut filter on SSB, rolling off (attenuating) audio below 100 Hz and even sometimes 150 Hz. On some SDRs this rolloff frequency can be adjusted. None-the-less it hardly matters, as a 1000 Hz chunk of the spectrum from 100 or 150 Hz to 3000 Hz is still available for analysis dependent on where you tune the receiver.

### **Choosing Proper Setup Parameters**

I will describe all the settings which make a difference, but not all spectrum analyzer programs enable you to set all of these. Pay attention to the ones you can set.

Sample rate decimation, FFT size, FFT span, input gain, AGC threshold, window overlap, waterfall scroll interval, waterfall brightness & contrast are the key parameters which are usually tweaked to give the best waterfall display from the weakest of signals. They all tend to interact, so it will take some additional experimenting on some of the more complicated spectrum analyzers like Spectrum Lab.

Strive for a final sample rate to the FFT of 6000 samples per second. Remember, you don't need a sample rate greater than 2x

the width of your receiver's bandpass filter (usually 3000 Hz maximum). Next, a good starting FFT size is 65536. If you can set the frequency span, set it initially to 200 Hz. This is the waterfall window width for a vertical waterfall, or the waterfall window height for a horizontal waterfall strip.

Keep your input gain under control and don't overload. Your input gain should fall somewhere near the AGC threshold, in dB. Avoid extremely low audio gains because they will introduce artifacts. Window overlap is best between about 50% - 90%. Finer waterfall detail will be found as you lower this value. Waterfall scroll intervals between 0.6 and 2.0 seconds are a good compromise for maintaining adequate waterfall detail and update speed. Keep smoothing at minimum. Smoothing will tend to smear signals. Finally, tweak brightness and contrast until you get the clearest display, bringing out the detail.

## **Viewing the Carrier**

The carrier in the waterfall at shortwave frequencies will be broad, meaning several cycles wide or more, and almost never a simple "pencil" line. This is due to ionospheric scattering over the long haul from transmitter to receiver. Mediumwave carriers, out to single hop distance anyway, will look more like a defined but slightly fuzzy line. Beware of perfectly looking straight pencil lines. They are most likely signal spurs coming from nearby electro-magnetic interference or a bad audio connection.

Doppler effects on carriers - slight drifting of two or three or more cycles over time - can be very interesting to watch. Catching a station as an auroral event starts, or the onset of solar wind from a solar storm does funny things to carrier traces on a waterfall. Watching a distant station as sunrise or sunset passes over that station (or over you!) will also cause doppler effects on its carrier as the ionosphere raises or lowers. Tune to time standard WWV & WWVH on 5.0, 10.0, or 15.0 MHz sometime when they are both in



and watch as sunrise or sunset over Hawaiian WWVH progresses. Even though they are precisely on the same frequency, you will see the two carriers split a couple of cycles apart as the ionosphere over WWVH changes height.

Earthquakes can cause doppler shifts in carrier frequencies. The recent 8.8 magnitude earthquake off the Russian Kamchatka Peninsula caused large doppler swings on carriers in the region (propagation monitoring sites 5006 kHz and 8006 kHz, Japan) and several VLF stations. If you are watching or recording at this time you will experience a thrill of a lifetime.

## **Things to Try**

Tune to Russian time standards station RWM in Moscow on 4996 kHz (5 kW), 9996 kHz (8 kW), or 14996 kHz (8 kW) at the right time of day or night and monitor its carrier. You may never actually log this station in a normal DXing scenario, but you can find it and see it using the carrier sleuthing technique.

The Chinese time station BPM at Xi'An, China broadcasts on 2.5, 5.0, 10.0, and 15.0 MHz, the same frequencies as WWV and WWVH. Check your sunrise and sunset maps for Xi'An and watch for it to show over WWV and WWVH at these times.

Investigate long path propagation. At certain times of year long path propagation will be enhanced, that is, signals arriving at your receiver location from the long way around the globe. Often both short path and long path are open and viewable at the same time. Long path tends to be best at sunrise and sunset and at the Spring and Fall equinoxes, but can be "seen" at any time of year. The Voice of Vietnam on 7435 kHz has been in via long path to western New York on a couple of days over the last two weeks (late July, 2025). Both short and long path signals were visible, separated by a couple of cycles due to the ionospheric height differences on the paths.

Watch the lower bands below 10 MHz during daytime hours through the winter season. These are normally considered nighttime bands. Due to lowered MUF (maximum usable frequency) many areas of the world will be receivable on shortwave in the 31, 41, and even 49 meter bands. During the winter season, daytime skywave DX will be possible on the mediumwave band as well. It is there, it is weak, and you can "see" it using carrier sleuthing techniques. You have the ability to "see" way more than you can copy audibly.

## **Schedules**

Who's transmitting and how do I find them? A number of shortwave schedules are currently produced twice a year and updated in between, too: [HFCC](#), [EiBi](#), [Aoki](#), and more. Consult these. Get familiar with them. There are websites which show SW schedules, most are derived from one of these lists. Here are a few:

[SHORT-WAVE-INFO](#)

[Shortwave.Live](#)

[Shortwave DB](#)

[ShortwaveSchedule.com](#)

A wealth of shortwave scheduling information is available on the web. Many stations also have a website or Facebook page.

The [MWLIST](#) website does about the best job documenting worldwide mediumwave stations. It has split the world into four zones. Within each zone is a page for each channel frequency, 530 - 1700 kHz.

[MWLIST Asia](#)

[MWLIST Europe, Africa, and Middle East](#)

[MWLIST North America, Central America, Caribbean](#)

[MWLIST South America](#)

## **Recording**

It is entirely possible to make audio recordings of your listening sessions, even for many hours while you are away or sleeping. Get your receiver set on the frequency of interest, set a timer to start the recording at a certain time (exactly on the hour preferably), and record the audio coming from the headphone output. A no loss .WAV file should do nicely, though a quality .MP3 format should work too.

At a later time or date you can use this recorded audio file as input to your spectrum analyzer and view the listening session. Be sure to set time stamps "on" at playback time so you can correlate the spectrum display with the actual recorded time.

Take care in naming your recorded audio file by including the frequency and start date-time in its name. That way you can refer back to it and know its origin.

Of course, I-Q recordings from an SDR can be saved and played back through the SDR receiver software to reproduce the audio file again too. Set your SDR to record I-Q data and record. Start your SDR receiver software and play back the I-Q file, routing the audio to the spectrum analyzer software.

## **Spectrum Lab Components Window**

Spectrum Lab can display a visual "map" of its systems (called Components), where you can connect amplifiers, modulators, wave generators, filters, recorders, and other frills to your waveform analysis. Once you get ramped up on the learning curve, enable Spectrum Lab's Components window and check it out.

## **Wrap Up**

For those a little more technically-inclined, Carrier Sleuthing can not only be interesting but bring renewed interest and excitement back into your radio hobby. It is very much like signals analysis in the Big Leagues: SIGINT, or Signals Intelligence. It folds right into a lifetime interest of mine, signal analysis.

As an add-on to this series, next we'll look into a neat little web-based propagation tool called Proppy. It will help you to know when to look for carriers and what might be possible to "see" at a given time of day and frequency on the shortwave bands.

Today you can still be a signal sleuth, and from the comfort of your home.