SpectraVue

SPECTRAL ANALYSIS PROGRAM

Document Version 1.21

MOETRONIX

www.moetronix.com

Table of Contents

1 Introduction

SpectraVue is a Windows based program that takes digitized signal data from a variety of sources and displays the frequency domain spectrum on the PC screen . It also can be used to demodulate various signals and output them to a soundcard or wave file. The input data can come from a soundcard(licensed version required), a RIFF .wav file, or from a special RF input capturing device, an SDR-14. (www.rfspace.com)

1.1 Functionality

Basic program functionality includes the following features:

- FFT analysis using 2048 to 262144 point FFT's (real or complex).
- FFT amplitude resolutions from 10dB to .2dB per division.
- FFT amplitude power is in dB referenced to full scale (a single sine wave with an amplitude of +/-32767 or 16 bits).
- Various FFT data view modes including waterfalls and 2D and 3D plots.
- Display markers for accurate amplitude/frequency measurement.
- A continuum display for measuring spectral power over the entire span and/or the peak power within the span.
- Frequency Spans from 5KHz to 30MHz in various step sizes. (Using SDR-14 unit)
- Signal demodulation of AM, FM, WFM, USB, LSB, narrow CW modes when using the 50 to 150KHz Frequency Span and SDR-14 unit.
- Programmable demodulation filters.
- Saving and playback of captured spectrum using RIFF .wav files.
- IF mode so program displays true frequency even if input is from a different IF frequency.
- A variety of file saving options ranging from saving the raw data to saving demodulated audio to saving various waterfall displays using graphics file format.

1.2 Example Uses

- General signal frequency domain analysis of spectra.
- Radio Astronomy applications where very weak wide bandwidth signals can be viewed with up to 30MHz bandwidth and a 254 Hz bandwidth resolution.
- Demodulating various signals and viewing signal activity over a 150KHz bandwidth in real time.
- Storing and playing back sampled data for later analysis or demodulation.

1.3 Getting Started

SpectraVue is a complicated program and it is suggested that the user familiarize ones self with the basic operation of the program by spending a few minutes reading through this help file and by playing with the various features of the program. A little time spent in this way will reduce the frustration factor of learning a new program and its controls.

1.3.1 System Requirements

In order to utilize all the features of SpectraVue a 1GHz Pentium 2 or better with 64M of memory is required. Much slower PCs can still be used if one is willing to accept slower displays and give up the real time demodulation features.

Windows XP, Win2K, or Windows 98SE is required.

The graphics card is probably the single most important item for maximizing the full potential of SpectraVue and the SDR-14. The faster the better.

In the FFT Setup menu there is a "Slow CPU Mode" spin control that the user can adjust to adapt slower graphics cards/CPUs for best results. The tradeoff is that the display will skip updates to conserve CPU time.

For simple slow spectral analysis, a 200 MHz Pentium 1 will suffice.

1.3.2 Installation

The SpectraVue distribution is contained in a single Setup.exe file. Execute this file from the CD or wherever it is located. The install program will run and step you through the process. One can choose alternate folders for the program and whether to automatically create a desktop icon and where to place the short cut in the Startup Menu.

The installer will create three folders under the SpectraVue main folder.

The \USBdriver folder contains the USB driver required by the SDR-14 hardware.

The \Palettes folder contains several .pal files that are optional palette files used for the waterfall colors.

The \FilterComp folder contain optional .fcf filter compensation files used for flattening out the SDR-14 filter shapes.

The two main files for the program are "SpectraVue.exe" and a single dll file "IOModule.dll".

Several .ini files are included with some program settings for a few common modes. The file Specravue.ini file is saved each time program exits and saves the users current settings.

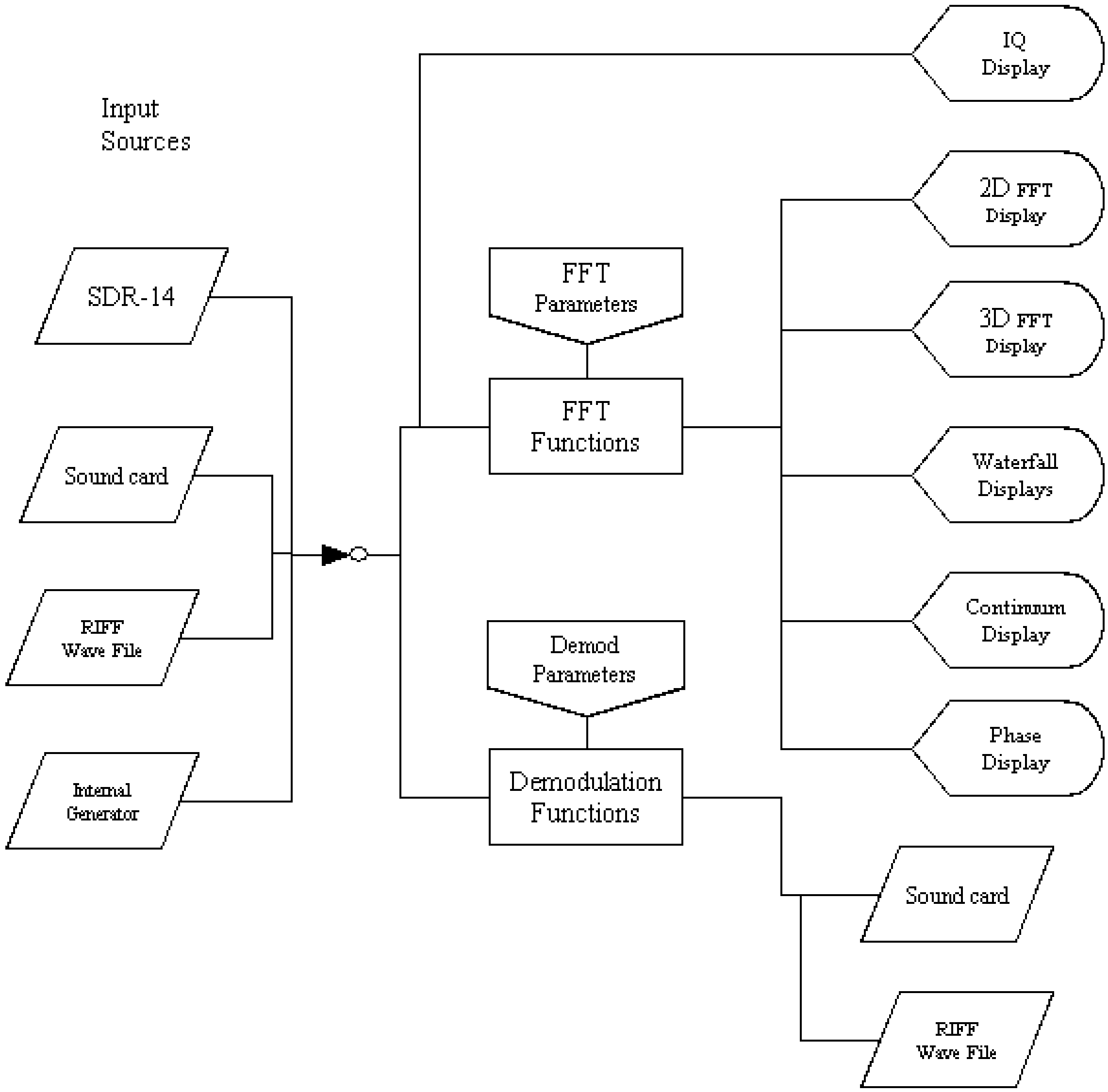
The "spectravue.chm" file is the online help file which can be executed directly or from within the SpectraVue program by clicking on Help.

1.3.3 Uninstall

The "unins000.exe" program can be used to uninstall SpectraVue from your computer or you can use the shortcut placed in the Windows Program/Start Menu. If any additional files were created during use such as .ini, .bmp, or .wav files, they will have to be manually deleted.

1.4 Program Overview

A block diagram of the overall program functions is shown below.



The data input can come from the USB port and an SDR-14 unit, a soundcard, a RIFF .wav file, or an internal test generator. A simple display of the raw data is available which is useful to set audio levels on a soundcard or adjust DC offsets.

The data stream is then sent to the FFT engine where the basic FFT is calculated as well as all the spectral peak and averaging functions.

A demodulation section may also process the data with the demodulated audio sent to a soundcard or wave file. There is more detailed information concerning the demodulator section elsewhere in this help file.

2 Program Details

Basic program setup and operation can be accomplished in the following order:

Select the desired Input Source. (Soundcard, wave file, or SDR14)

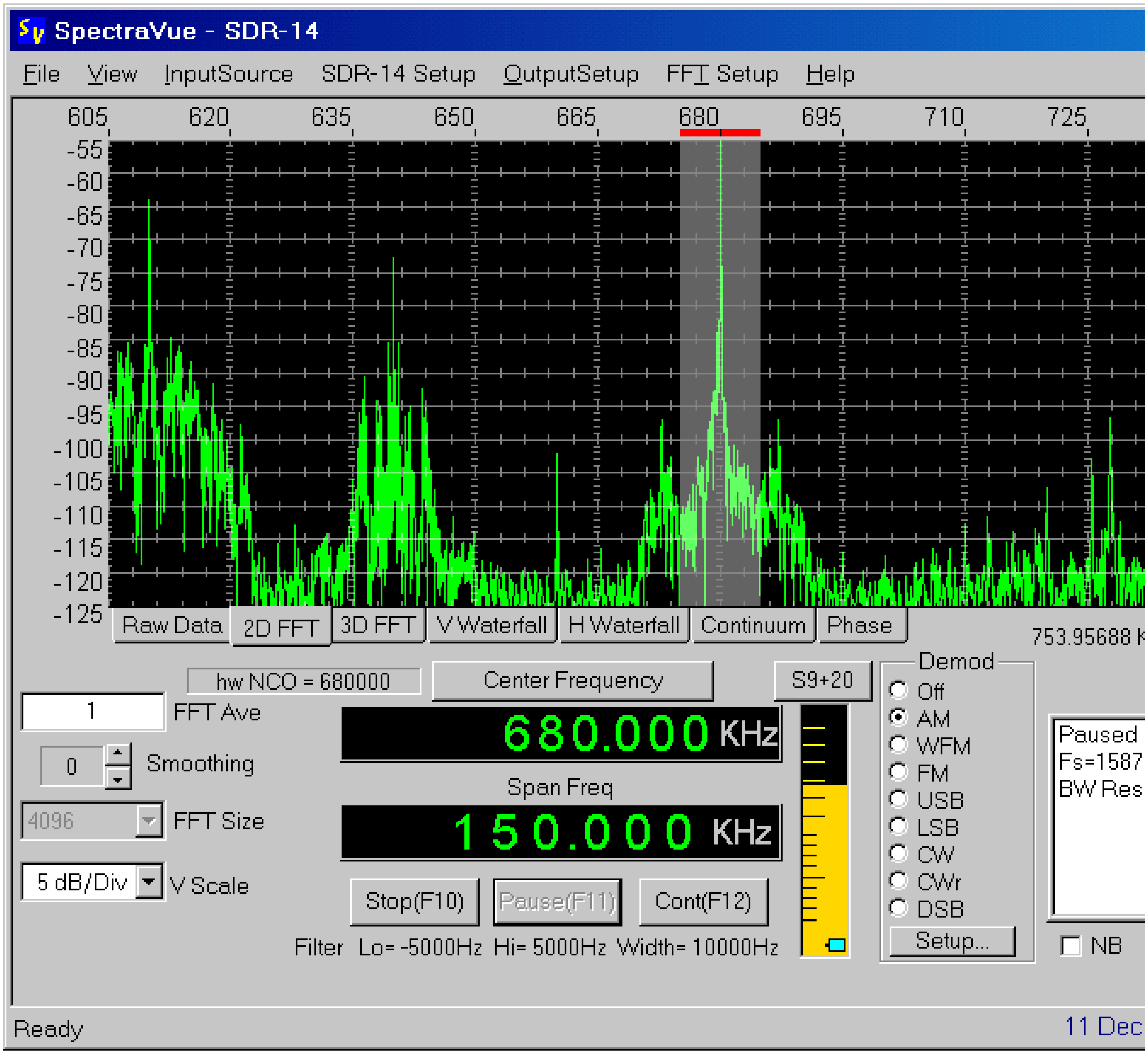
Setup the selected Input source. (Select bandwidth, gains, etc.)

Select Output Setup and chose the soundcard, wave file or other output modes.

Select FFT Setup and select any display options desired.

On the main screen select the FFT parameters desired. (Size and averaging)

On the main screen press the START button to begin capturing.



Across the top of the screen are menu buttons that are used to setup the majority of all the program settings.

The main screen of SpectraVue contains the primary signal viewing area. Tabs below this area allow the user to select display variations. Two frequency controls are below the tabs for setting the center frequency, demodulation frequency, and frequency span. Below these controls are the controls to start, stop or pause the capturing process.

On the lower left are controls for adjusting the several FFT parameters. On the lower right are the demodulation selection controls and also a general information box that show status and other information.

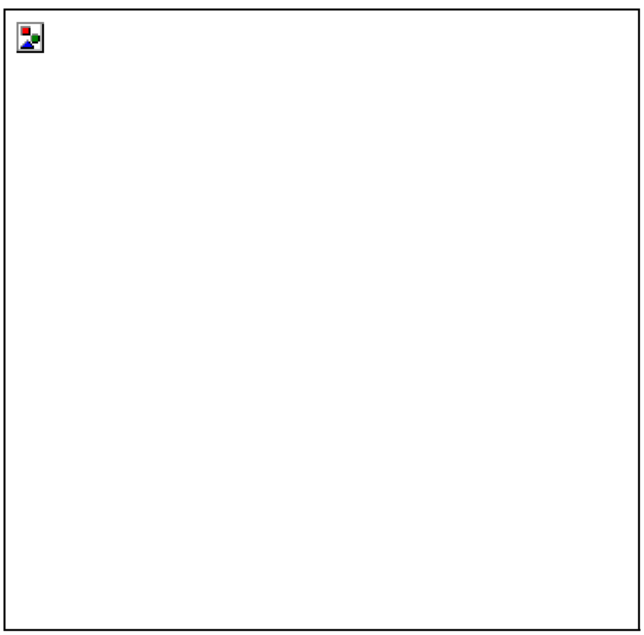
A slider control to the right of the main viewing screen is used to move the viewing window up and down to place the displayed signal into view. The Auto Scale button will try and center the display automatically.

2.1 Screen Views

The main program contains a tabbed view screen where the user can click on a desired display mode tab. There are seven different view screens that can be selected.

2.1.1 Raw Data

This view is of the raw input data versus time. Its purpose is to give the user a quick look at incoming data to verify its general amplitude. There are no user controls to this display. The amplitude is automatically scaled to fit the screen and the time axis is not calibrated but is a function of the sample rate and screen resolution.



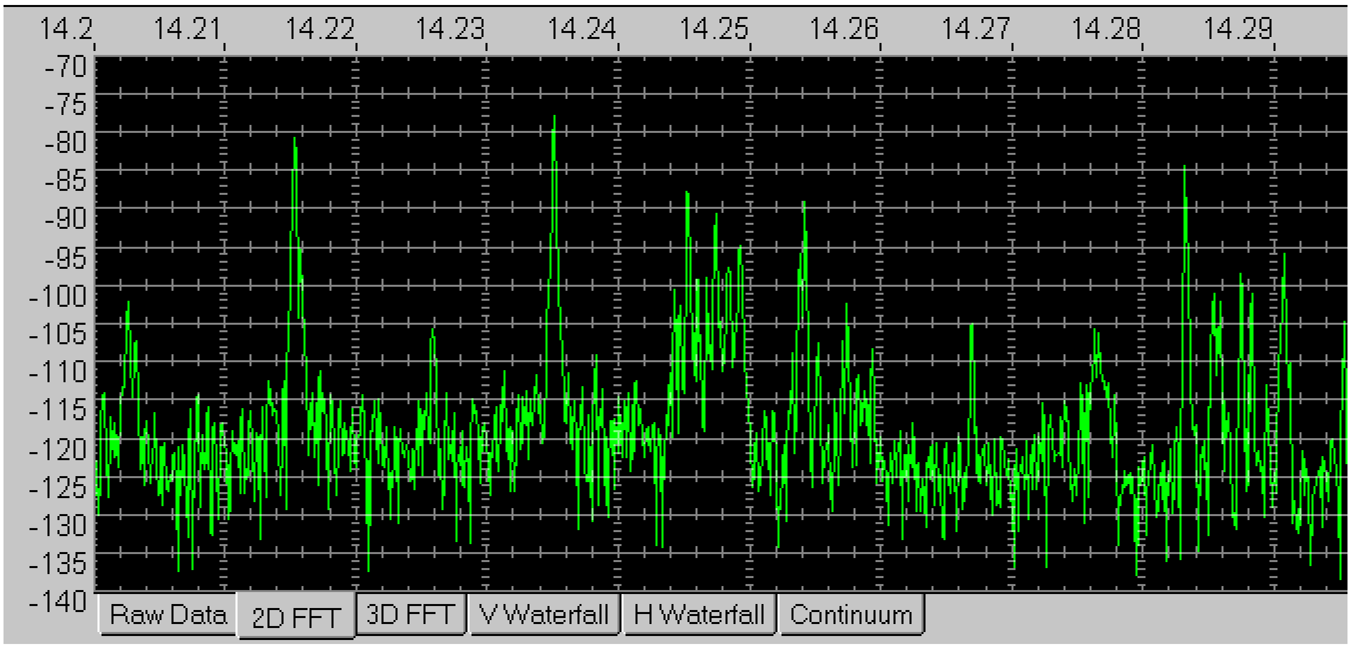
If it is real data then there is only one trace. Complex data is displayed as two different traces with different colors. The two numbers in the display area are the max and min values of the incoming signal. The range of inputs is a 16-bit value or 0 to +/- 32767.

The DC offset is displayed to the left of the screen. This may be of use for adjusting the soundcard DC offset input parameters.

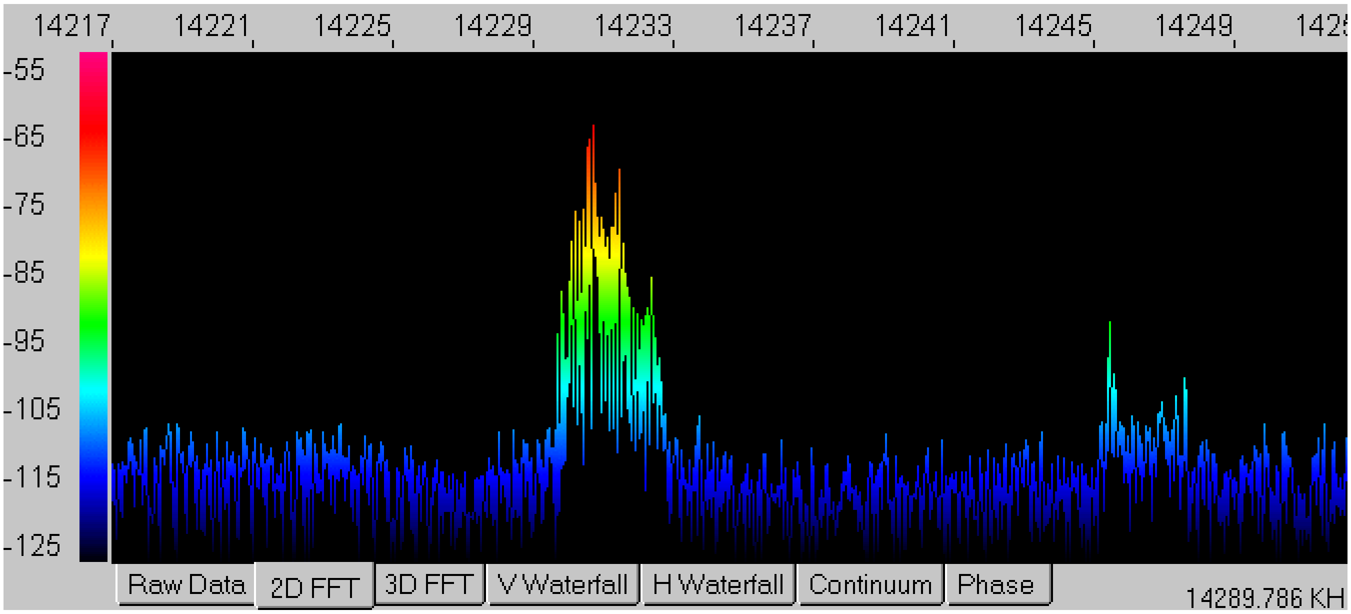
2.1.2 2D FFT

The 2D FFT display shows the spectrum amplitude versus frequency plot of the incoming signal. The center frequency and frequency span are set by the two main controls just below the FFT display area. Amplitude in dB referenced to full scale is displayed on the left side of the display and frequency along the top. The slider control on the right can be used to shift the amplitude display up or down.

> FFT amplitude power is in dB referenced to full scale (a single sine wave with an amplitude of +/-32767 or 16 bits). This can be calibrated to dBm by entering a dB offset in the FFT Setup menu.

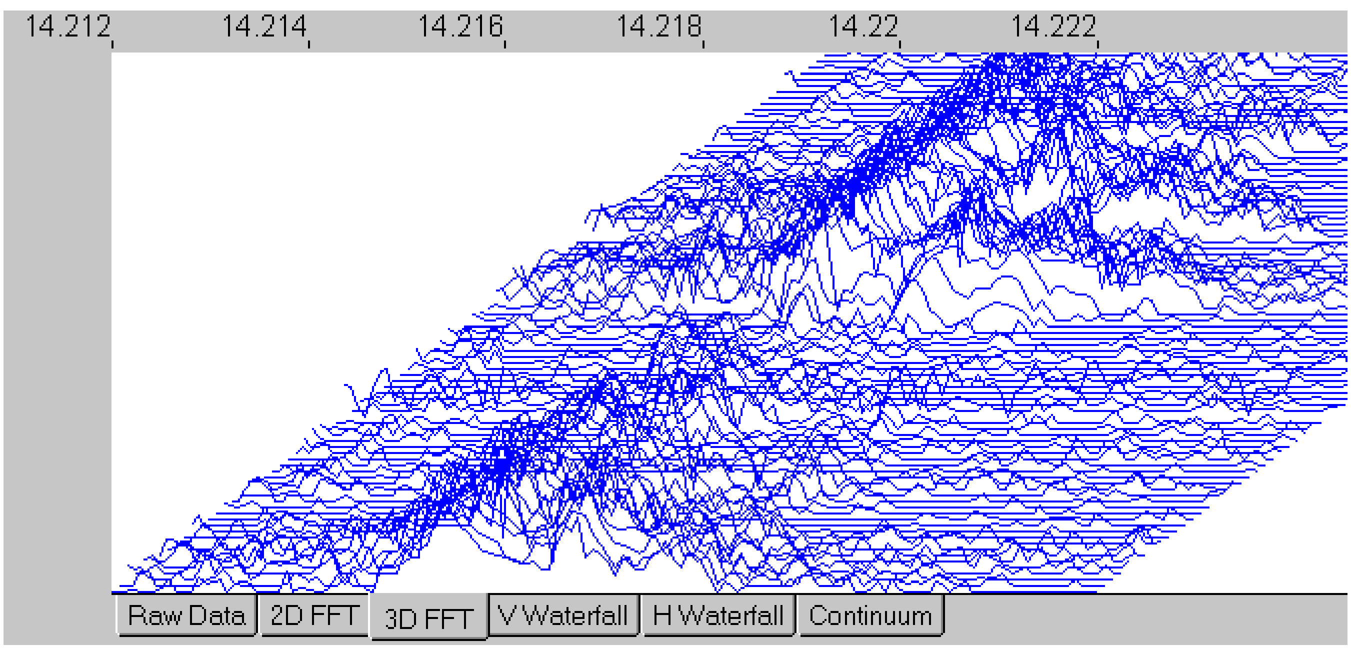


An alternative 2D view is available by choosing the "Color 2D Graph" selection in the FFT Setup Menu. This view adds color to the display in place of the internal scale markings. The colors correspond to the Waterfall color palette.



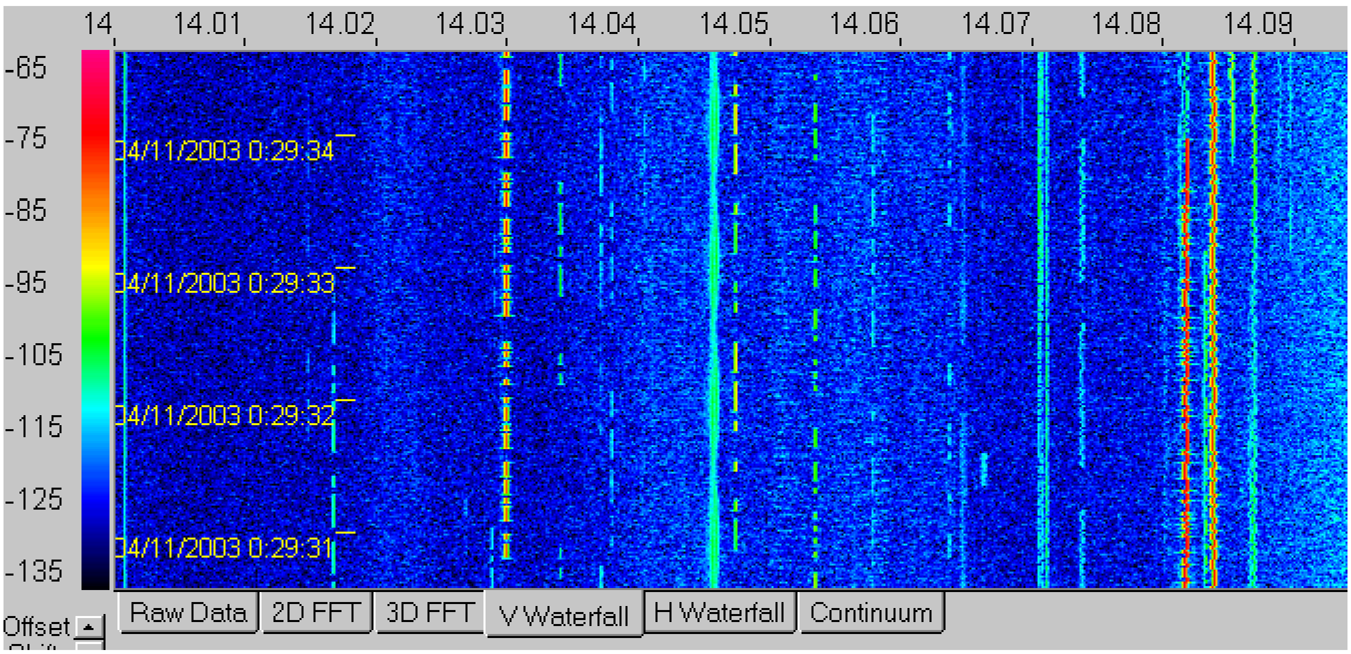
2.1.3 3D FFT

The 3D FFT display shows the spectrum amplitude versus frequency plot of the incoming signal and then provides a running history in time of the FFT plot by shifting each past display up and to the right. The center frequency and frequency span are set by the two main controls just below the FFT display area. Amplitude in dB referenced to full scale is displayed on the left side of the display and frequency along the top. The slider control on the right can be used to shift the amplitude display up or down. For better displays, play with the amplitude offset bar on the right and also the FFT smoothing value.



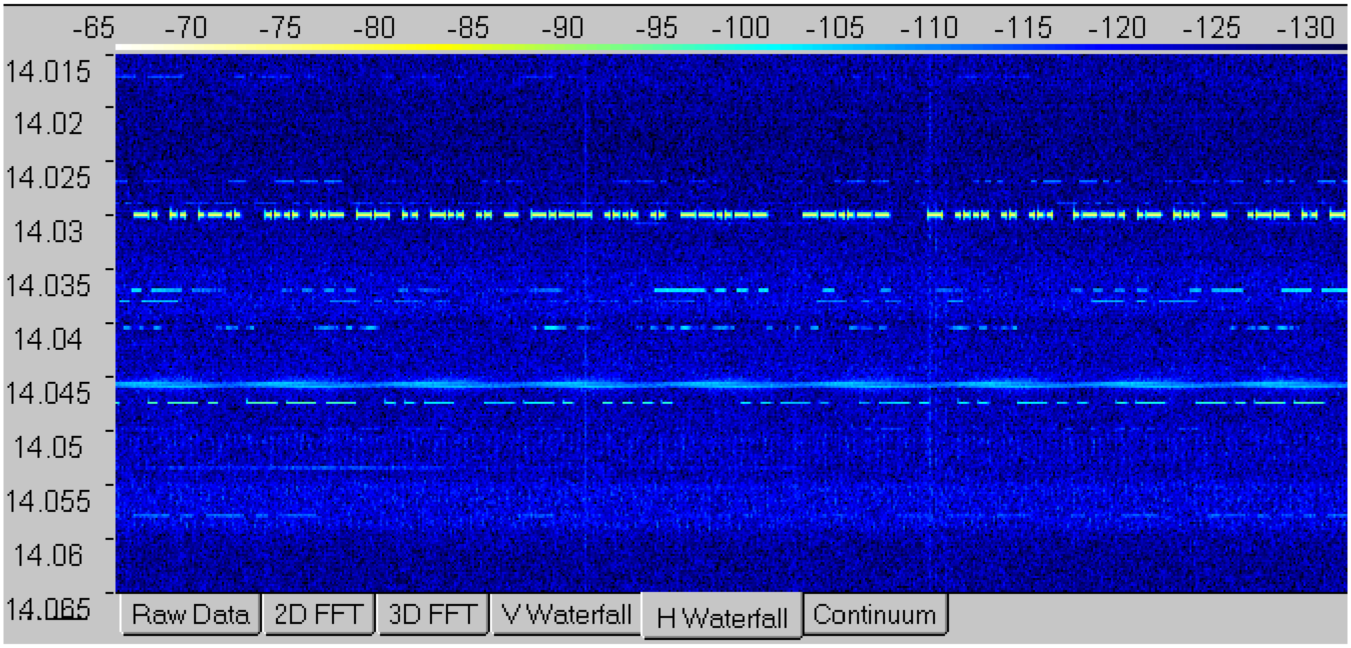
2.1.4 V Waterfall

The vertical waterfall FFT display shows the spectrum amplitude as a color instead of as a height on the display. Frequency is the horizontal axis and time is the vertical axis. The center frequency and frequency span are set by the two main controls just below the FFT display area. Amplitude in dB referenced to full scale is displayed on the left side as a color scale. The slider control on the right can be used to shift the amplitude display up or down in large increments and the spin control on the lower left side of the display can be used to make small shifts in the color amplitude map. An optional timestamp feature can be invoked(from the FFT Setup Menu) to mark the UTC time as the display scrolls down.



2.1.5 H Waterfall

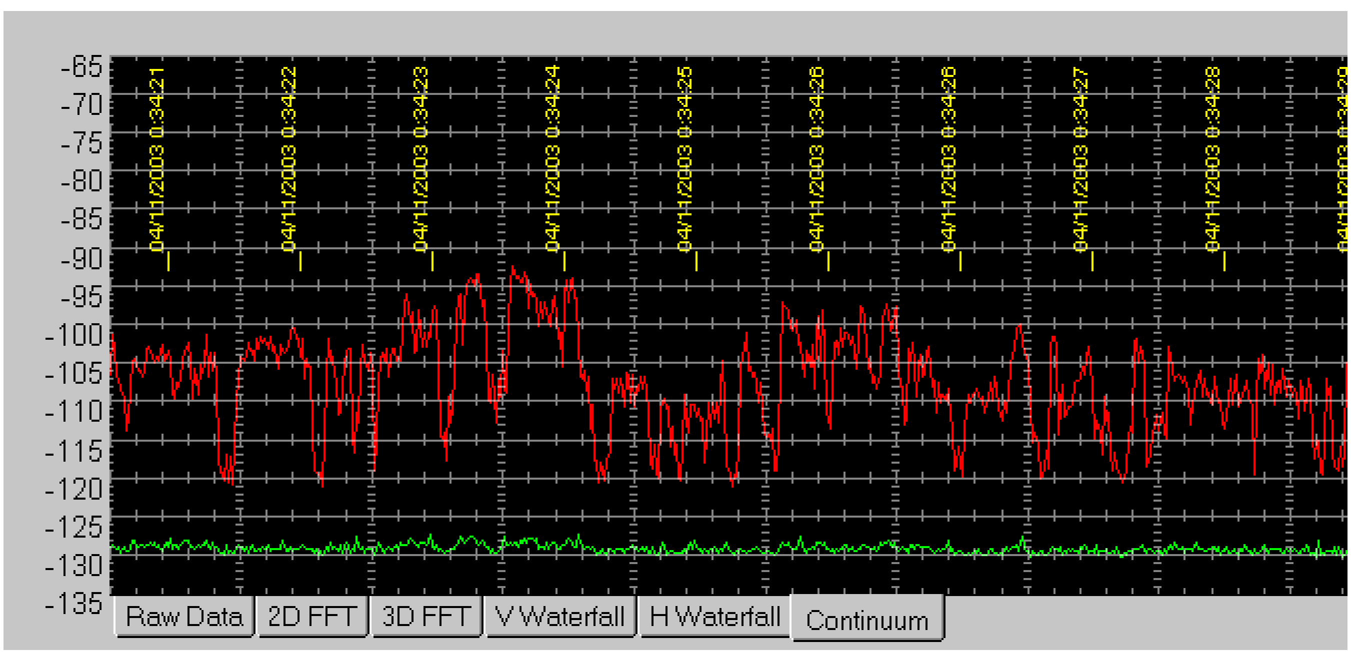
The horizontal waterfall FFT display shows the spectrum amplitude as a color instead of as a height on the display. Frequency is the vertical axis and time is the horizontal axis. The center frequency and frequency span are set by the two main controls just below the FFT display area. Amplitude in dB referenced to full scale is displayed on the topside as a color scale. The slider control on the right can be used to shift the amplitude display up or down in large increments. An optional timestamp feature can be invoked(from the FFT Setup Menu) to mark the UTC time as the display scrolls left.



2.1.6 Continuum

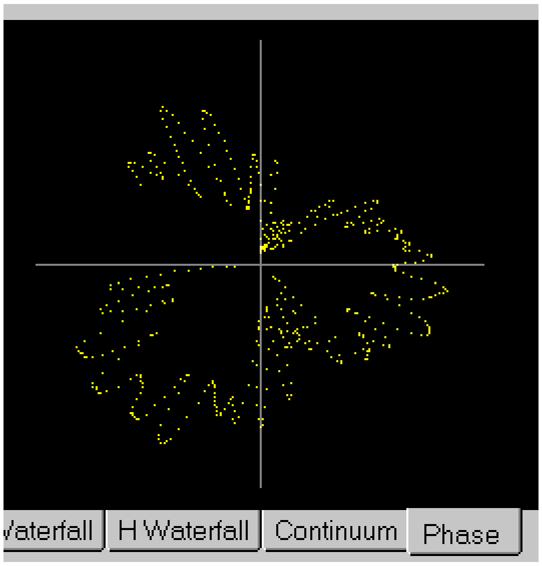
The continuum display mode displays the average power over the entire frequency span versus time. An optional secondary trace can be displayed that also shows the peak power within the same span versus time. This is invoked from the FFT setup menu by selecting the Enable Peaks check box. Time stamping can also be enabled(from the FFT Setup Menu) for this scrolling display.

FFT amplitude power is in dB referenced to full scale (a single sine wave with an amplitude of +/-32767 or 16 bits). This can be calibrated to say dBm by entering a dB offset in the FFT Setup menu.



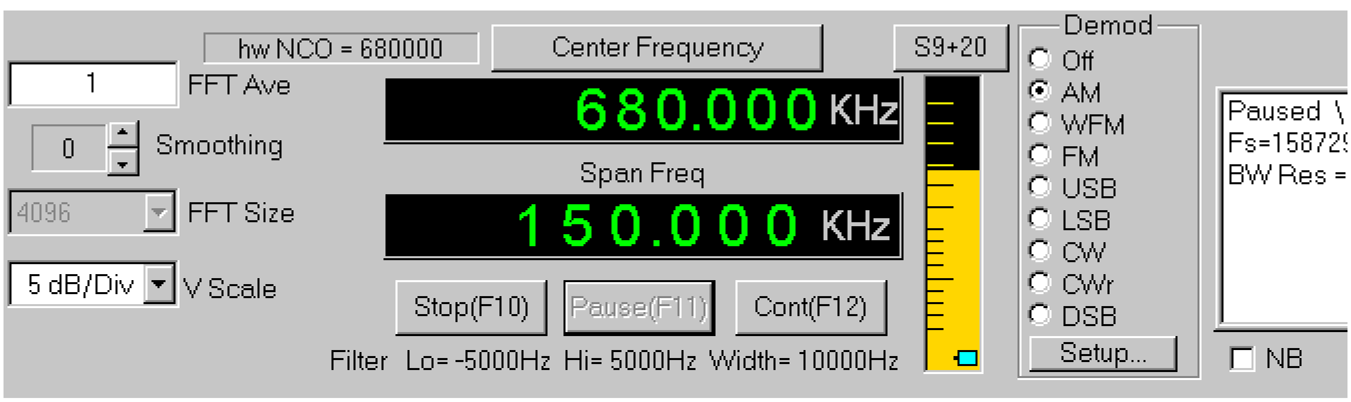
2.1.7 Phase

The Phase display mode displays the I and Q data points in an x-y axis plot. 512 points are plotted at a time. The signal strength must be fairly strong to give interesting results and also as narrow a bandwidth so multiple signals don't confuse the display.



2.2 Controls

Various controls are available on the bottom of the main screen to set frequencies, FFT size, and other common settings.



2.2.1 Center Frequency Control

The center frequency control is used to specify the display center frequency. This value and the frequency span value determine the FFT display's range and position within the FFT. If the SDR-14 is used as an input source, this frequency is also the NCO frequency that corresponds to the zero frequency of the complex FFT. The frequency units can be specified in the FFT setup menu.

The frequency control can be set in several ways.

Place the mouse cursor on the digit you wish to change or press one of the arrow keys. The digit background will change color. Use the left or right arrows to change digit positions within the control.

- Type the desired numeric digit using the keyboard. The selected digit will move to the right so that an entire frequency value can simply be entered from the keyboard.

- Click the mouse on the top of the selected digit to increment that digit value or on the bottom of the digit to decrement it.

- Use the up or down arrows to increment or decrement that digit value and zero all following digits.

- Use the Page Up or Page Down keys to increment or decrement just the selected digit and not the entire frequency value. (digit value does not roll over to adjacent digits)

- Click the left mouse button within the display area on the frequency you wish to center.

- A USB "Power Mate" knob from Griffin Technologies can be mapped to the following keys: http://www.griffintechnology.com/

CTRL-F1 = digit decrement (Rotate Left)

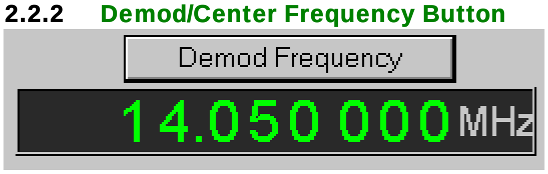
CTRL-F2 = digit increment (Rotate-Right)

CTRL-F3 = move digit position left (Click and Rotate Left)

CTRL-F4 = move digit position right (Click and Rotate Right)

CTRL-F5 = Toggle frequency control between center and demod function.(Actually a "double-click" operation is required.(Click)

2.2.2 Demod/Center Frequency Button



The Center frequency control has a dual function when running the program in the demodulation mode. This button sets the function of the control to either the normal center frequency control or it can be used to set the demodulation center frequency which can be anywhere within the display frequency span. This button is only active if in the demodulation mode.

When in Center Frequency mode, clicking on a position of the screen will make that position the new center frequency.

When in the Demo mode, clicking on a position of the screen will make that position the new demodulator frequency and keep the center frequency fixed.

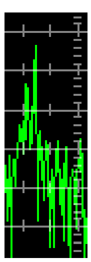
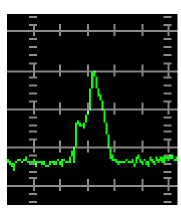
2.2.3 Span Frequency

The span frequency sets the range of frequencies that will be displayed on the screen. The display will always show frequencies from Center - Span/2 to Center + Span/2. The control operates exactly as the center frequency control except the arrow keys do not automatically invoke the control like the main frequency control.

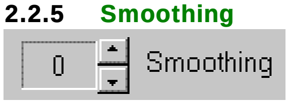
2.2.4 FFT Ave

The FFT average control is used to specify an averaging function that is applied to the FFTs. This is a point by point running average up until the number of samples reaches the specified average limit then it becomes a low pas filter with a time constant approximately equal to the averaging value.

This function is useful in averaging out noise and allowing a signal to be more easily viewed in a noisy background. The down side is that fast moving signals will become blurred.

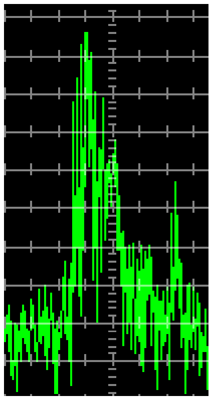
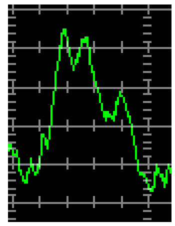
No average  average=100 

2.2.5 Smoothing

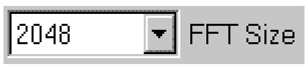


The smoothing function averages FFT bins within the same sample data block. This function is useful for wide bandwidth signals and smoothes the overall shape of a wide signal.

(Note: this is very CPU intensive so can slow down things if large values are used)

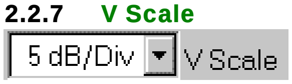
No smoothing  Smoothing = 6 

2.2.6 FFT Size



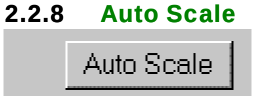
The FFT size pull down selection control is used to select the size of the FFT. The range is from 2048 to 262144 points. The larger the FFT the better the frequency resolution but the longer it will take to process since more data points are required between updates.

2.2.7 V Scale



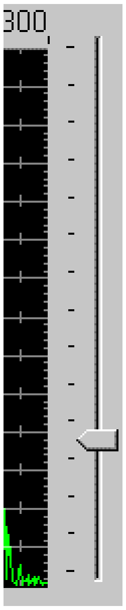
The Vertical scale control allows the selection of dB per division for the FFT amplitude. The range is from .2dB/Div to 10dB/Div. The display will only show 14 divisions at a time so the slider control on the right side of the display can be used to change the viewable range of amplitudes.

2.2.8 Auto Scale



This button can be used to perform an automatic centering of the signal within the display area. It performs an average of all the FFT points and adjusts the vertical position so that the average is somewhere near the bottom of the screen. This function is automatically invoked whenever the vertical resolution is changed. This control is particularly useful in finding the display trace if it is off screen.

2.2.9 Manual Vertical Slide



This control is used to move the viewable vertical window of the display. The control can be changed by dragging the control arrow or by clicking on either side of the arrow. The total range of the display is -140 to 140 dB referenced to full scale.

2.2.10 Start/Stop/Pause/Cont

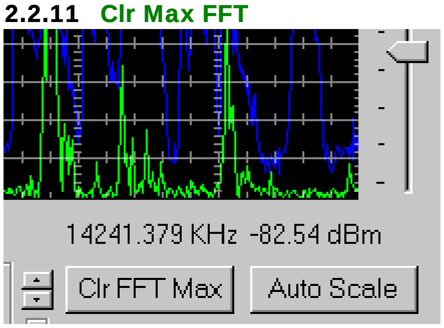
Three buttons at the bottom of the screen control the starting, stopping, pausing, or resuming of the data capture process. Function keys F10, F11, and F12 are mapped to these buttons as well.

Stopped State:

Running State:

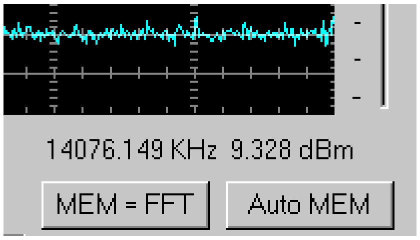
Paused State:

2.2.11 Clr Max FFT



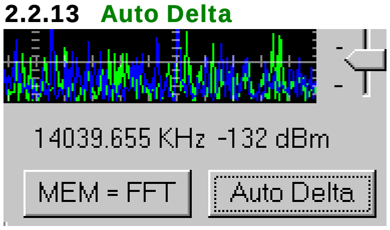
This button is used to reset the max hold buffer when the maximum hold display is active.

2.2.12 MEM=FFT



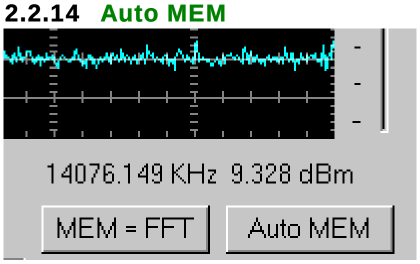
This button is used to store the current FFT display into the reference buffer when the delta FFT mode is active.

2.2.13 Auto Delta



This control auto scales the display to show the FFT delta buffer.

2.2.14 Auto MEM



This control auto scales the display to show the FFT stored memory buffer.

2.2.15 NB

This control toggles the Noise Blanker on and off. The noise blanker threshold and a duplicate on/off control are in the Demod Setup Menu.

2.3 Setup Menus

Various setup menus are accessible along the top of the program screen.

2.3.1 File Menu

2.3.1.1 Load/Save Setup

These menu items allow the user to save all the program setup information into an .ini file that can be uniquely named and loaded into the program later to restore the particular settings saved in the file. This allows the user to create a library of various custom setups that can then be quickly loaded without having to setup all the various options again. The most recently used file list can be clicked on to recall the last several ini files that have been used.

Note that the file Spectravue.ini is the primary file for saving your settings when exiting the program and will be overwritten each time the program is exited. It is loaded automatically when the program is first executed.

Note: This functionality is not available in the SpectraVue Evaluation version. A valid License file must be purchased to enable the user settings features.

2.3.1.2 Save FFT to Excel File

This menu item allows the current raw FFT data to be saved to a comma-separated file that can be imported into a spreadsheet for further analysis or graphing. The file contains 2 columns. The first column is the frequency in Hz and the second is the amplitude in dB relative to full scale. This only saves one FFT block of data th at was the last one captured.

2.3.1.3 Save Screen Graphics File

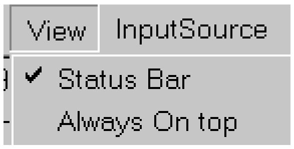
This menu item allows the current screen to be saved to a bitmap file. The format is an uncompressed .BMP file. Make sure the capturing process has been stopped before capturing.

2.3.1.4 Print/Print Setup/Print Preview

These menu items are used to invoke the Windows printer functions. The current screen can be printed out. One must stop or pause the capturing process before printing or the screen may not print as expected.

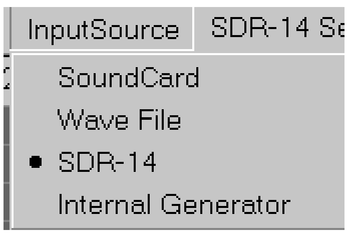
2.3.2 View Menu

The view menu allows the user to hide the bottom status bar and/or force the program to always be on top of the desktop.



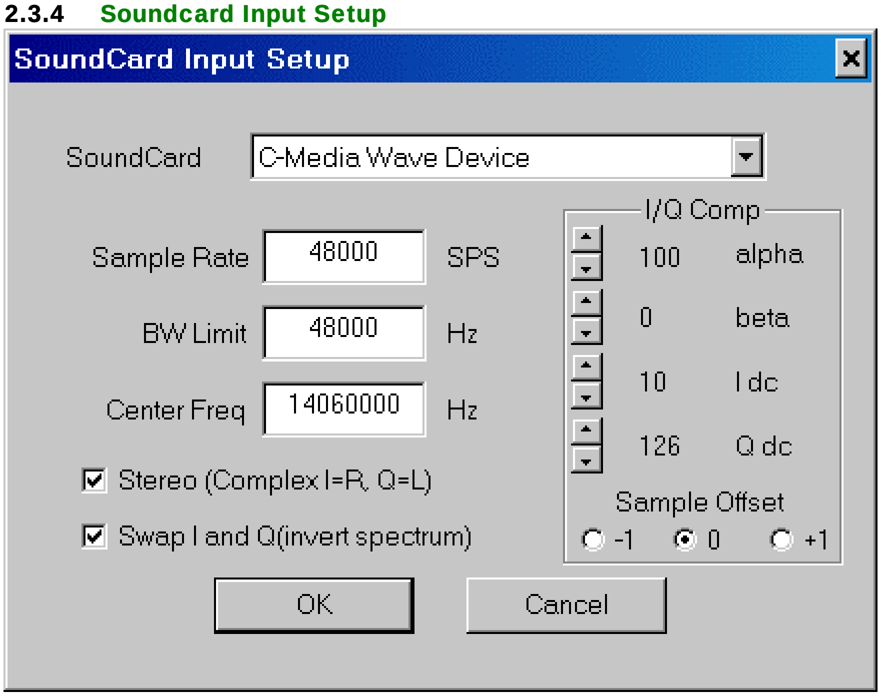
2.3.3 Input Source

This menu item allows the selection of a data input source. Currently 4 sources can be used.



The Input setup menu depends upon the input source selected.

2.3.4 Soundcard Input Setup



- The Soundcard can be selected with the pull down menu or one can use the default Windows soundcard.

- The sample rate must be specified and can range from 5000 to 96000 SPS.

- The bandwidth limit depends upon whether the Stereo (Complex) mode is chosen. The maximum BW is the sample rate if Complex or 1/2 the sample rate it mono (real) data.

- The Center frequency is an offset that is applied to the display. This is useful in applications where the soundcard is used to process base band I/Q inputs from complex mixers. The Center frequency would be the mixer oscillator frequency.

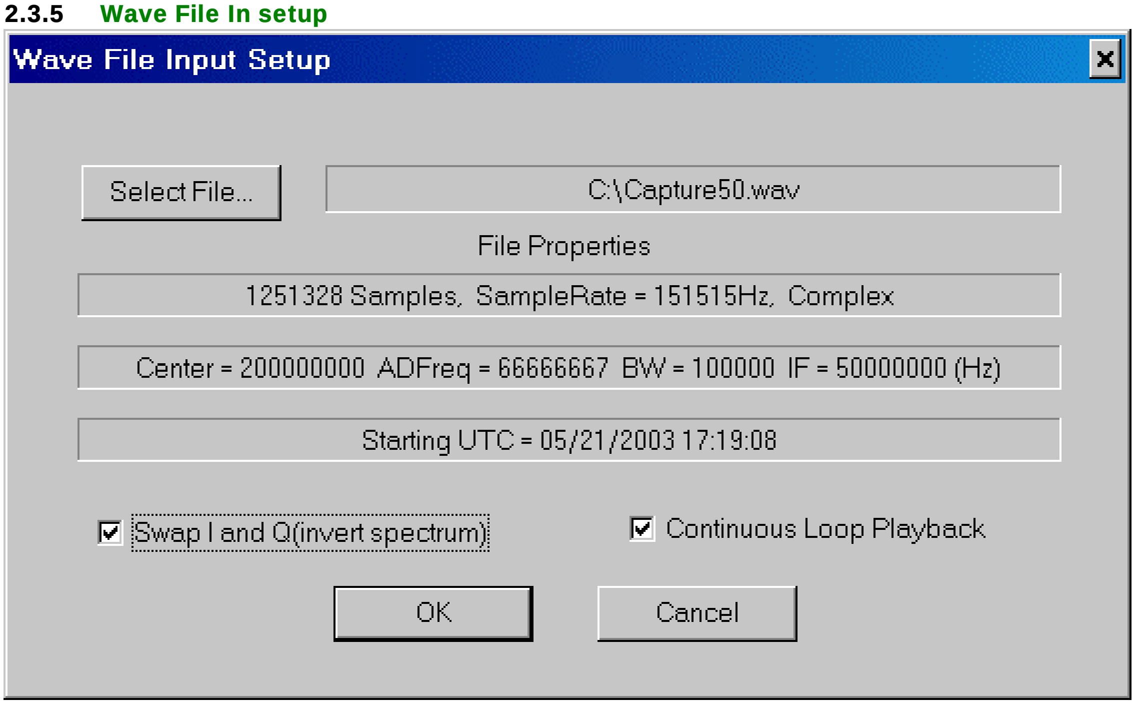
- The Swap I/Q button can swap the left and right channels that will invert the spectrum.

- The I/Q Comp section is used to adjust for DC offsets in the I/Q channels. By watching the raw data display (center number is DC offset) one can minimize the DC offset. The affect of a DC offset is a large spur in the center of the 2D FFT display.

- The alpha and beta controls can be used to compensate for any phase errors between the I and Q channels. The affect of a phase error is a reduction in sideband rejection. (A carrier will appear mirrored on the opposite side of the FFT display)

- The Sample Offset is a gross adjustment in case a soundcard is off by a complete sample between the I and Q channels. (One USB soundcard chip has a bug so that I and Q are one sample time apart.)

2.3.5 Wave File In setup

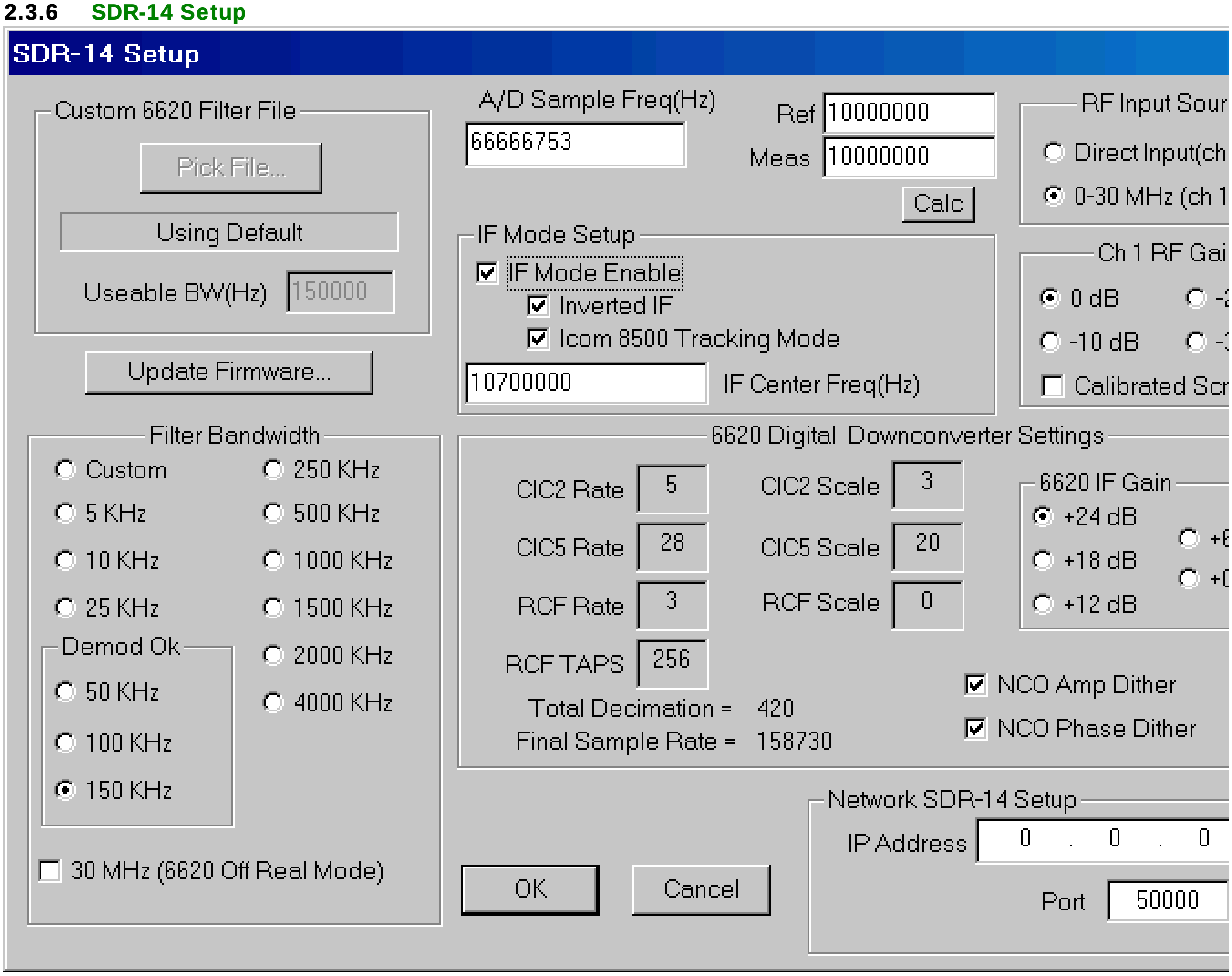


- The Wave file input menu allows the user to specify the wave file path and also whether to invert the spectrum or play the file back in a continuous loop.

- Depending on the file, several parameters are displayed along with the name. The first line shows the file size in samples, the file sample rate, and whether the data is real or complex.
- The second line may show the captured data center frequency, the A/D frequency, and display bandwidth if the SDR-14 was used to generate the wave file.
- The third line may show the capture starting time if the file was created with SpectraVue. This is used to show the original UTC time when playing back wave files. Note that if a capture was paused or the center frequency changed during creation, then the file data will not be correct for the entire file.

Wave files created with a program other than SpectraVue will not contain some of the above information.

2.3.6 SDR-14 Setup



This setup menu is used to set the main parameters of the SDR -14 hardware.

- The A/D Sample Frequency can be entered here. Nominally it is 66,666,667Hz but this number can be changed to the actual oscillator frequency if more accurate frequency measurements are desired. Entering a reference frequency and a measured frequency and pressing the "Calc" button will calibrate the sample frequency automatically.

- There are 13 fixed bandwidth setups that can be chosen along with a custom setting where the user can supply their own AD6620 setup data. If the 30MHz bandwidth is selected, the AD6620 down converter is bypassed and the entire 33.3 MHz wide A/D data is directly processed by SpectraVue using a real mode FFT.

Note that for demodulation, only the 50KHzto 150KHz bandwidths can be used. If the bandwidth menu is grayed out, make sure that Demodulation is disabled on the main screen menu.

- The AD6620 settings area of the screen displays the current AD6620 decimation rates and filter sizes. The AD6620 IF gain can be selected from 0 to +24dB in 6dB steps. This value is best kept at +24dB unless very strong signals are being analyzed and the SDR-14 is being overloaded.

- The input source can be selected as either direct to the A/D (ch0) or through a preamp and a 30MHz low pass filter (ch1).

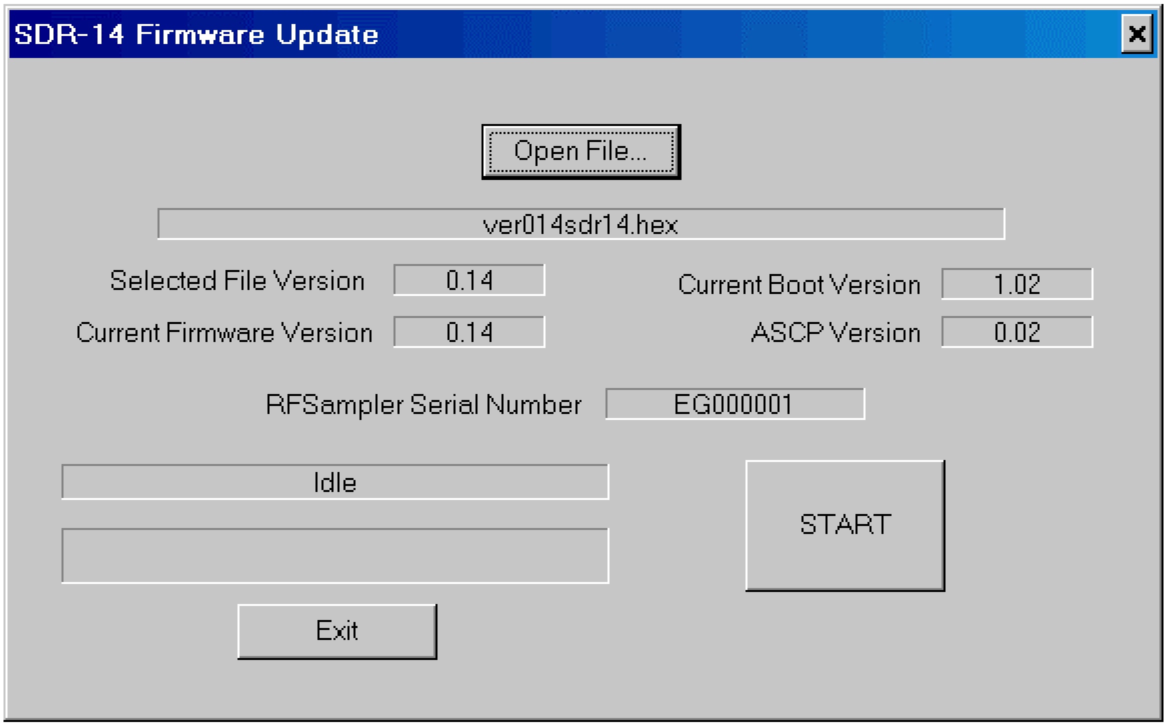
- If ch1 is used, the preamp/attenuator RF gain can be selected. The "Calibrated screen" check box forces the display to remain calibrated regardless of RF gain setting.

- An IF mode allows one to specify the IF center frequency and whether the spectrum is inverted. In this mode the SDR14 stays fixed on the IF center frequency entered here. The Center Frequency display is used to enter the actual external receive frequency. If the ICOM 8500 tracking mode is selected and an optional aux port to serial port adapter is used between the SDR14 and the ICOM 8500, then the radio and SpectraVue center frequency will automatically track.

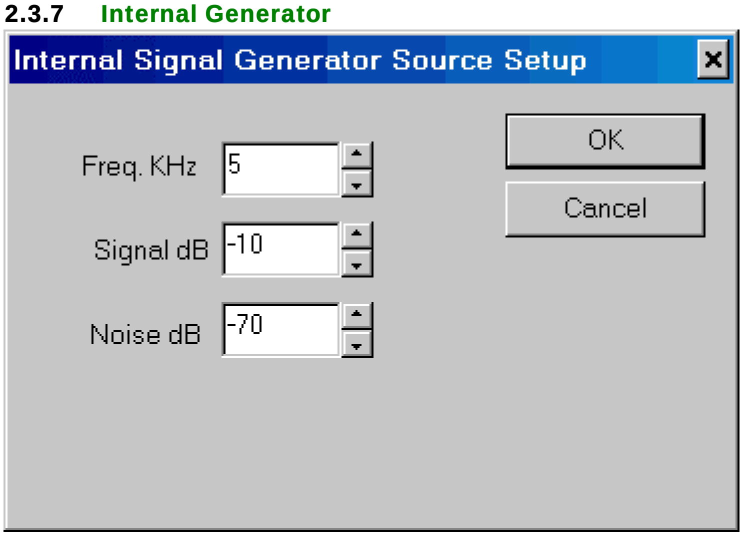
- AD6620 dithering can be enabled or disabled. There is no noticeable affect of these controls.

- If the Network SDR14 connection mode is enabled, the edit boxes for the desired IP and port addresses will be available. This feature is an option and requires a separate USB to TCP/IP Server application running to be able to use.

- The Firmware update button enables a menu to update the SDR-14's internal firmware. One opens the update .hex file and presses Start to begin the firmware update. The SDR-14 must be attached and powered up for this menu to be used.



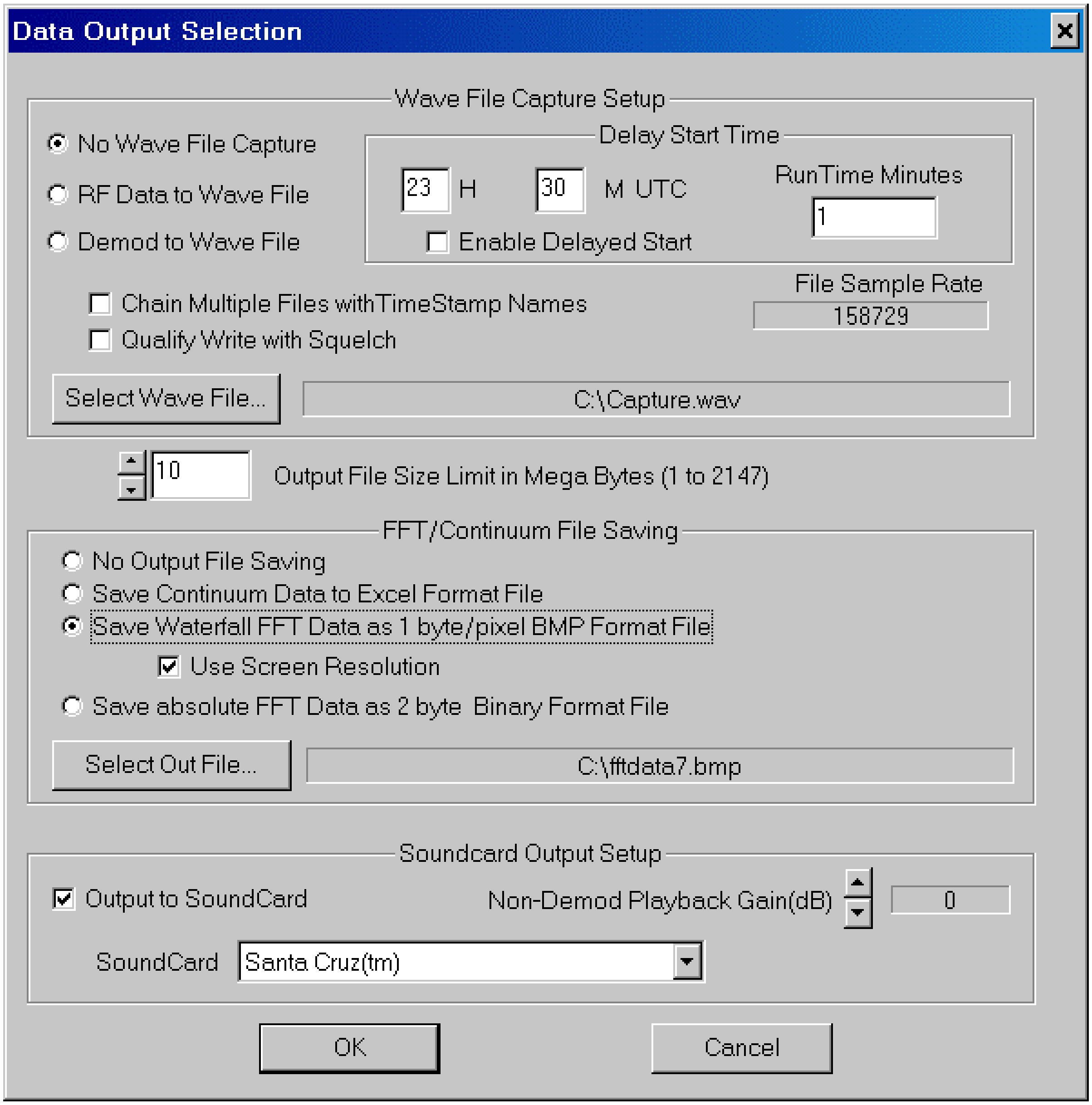
2.3.7 Internal Generator



This menu is used to set the internal test generator signal amplitude, frequency and noise source levels. This is a simple real mode sine wave generator running at 48KHz with a Gaussian noise generator that can be added to the signal.

2.3.8 Output Setup

This menu selects various output modes for the program.



2.3.8.1 Wave File Capture Setup

The top section sets up a wave file as an output destination. There are two formats that can be used. One is the I/Q (or real) RF Data can be captured directly to a wave file and then later used as an input source to this program. The other format is Demodulated real data that can be saved to a wave file. This is only valid if demodulation is valid.

The File sample rate is displayed to allow one to estimate the required file size. It takes 4 bytes per sample if the data is complex, 2 bytes per sample if real.

An edit box and spin control can be used to specify a maximum output file limit. The Range is from 1 to 2147 Megabytes. This is applied to all output files regardless of type.

2.3.8.2 Wave File Chaining

This feature allows very long data captures by breaking the output wave files into multiple files.

The checkbox for chaining multiple files enables a mode where SpectraVue will write to a wave file until the maximum file size limit is reached and then close that file and start writing to a new file. These filenames are created from the name provided by the user with a time and date label added to the end of each filename.

For example if the user chooses a filename of "test.wav" and the chaining feature is enabled, the filename will be changed to "test_YYMMDD_HHMMSS.wav" where YY is the last 2 digits of the year, MM is the month, DD is the day, HH is the hour, MM is the minute, and SS is the second. All times and dates are 24 hour UTC time.

The file chaining will continue indefinitely so it is up to the user to make sure enough drive space is available. It is a good idea to set the file size limit as big as you can deal with to prevent a large number of files being produced. Perhaps 640 Megs or so would be a good number so it could easily be archived to a CD.

The SDR-14 can create very large files very quickly so it is a good idea to be aware of how much space is available and how fast SpectraVue can create large files.

2.3.8.3 Qualify Write With Squelch

If the demodulator is enabled, then the "Qualify Write with Squelch" check box will only save to the file when the squelch control is open. This is useful for capturing to disk only the active time of a signal.

Note that the timestamp information in the file will not be accurate since the file is not being written to continuously.

2.3.8.4 Delayed Start Function

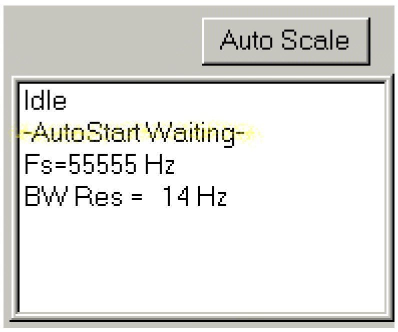
SpectraVue captures can be automatically started at a user-defined time for unattended operation. This is most useful for capturing data to wave files late at night or when there is no one around to start it.

Two parameters must be set. One is the start time in hours and minutes. This allows SpectraVue to be started up to 24 hours from the current time.

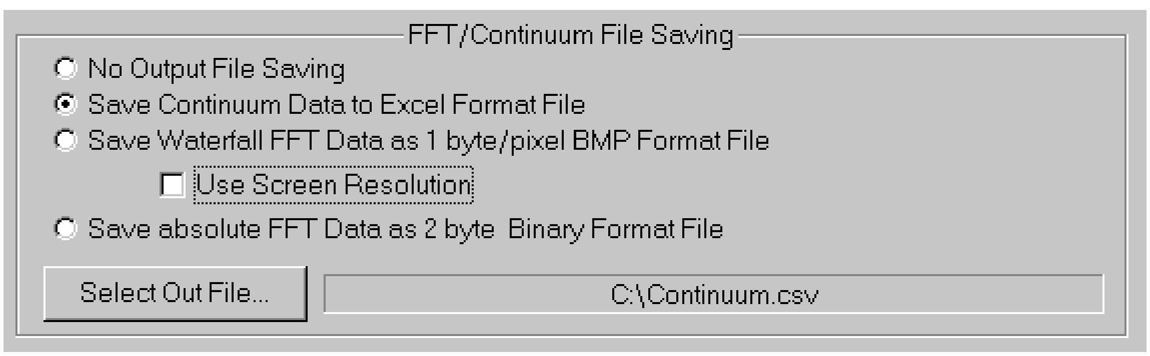
The other parameter is the length of time in minutes that is desired to be captured. The minimum time is 1 minute; the maximum time is 1440 minutes (24hours). Note that if not using the file chaining mode, the capture will stop when the file size is reached or the specified run time, whichever occurs first.

To enable the Automatic delayed start mode, the check box MUST be checked. This check box will be cleared upon program exit or after a previous run.

If enabled and waiting or running, the status will be shown in the "Info Box" on the main screen.



2.3.8.5 Save Continuum Data to Excel Format File



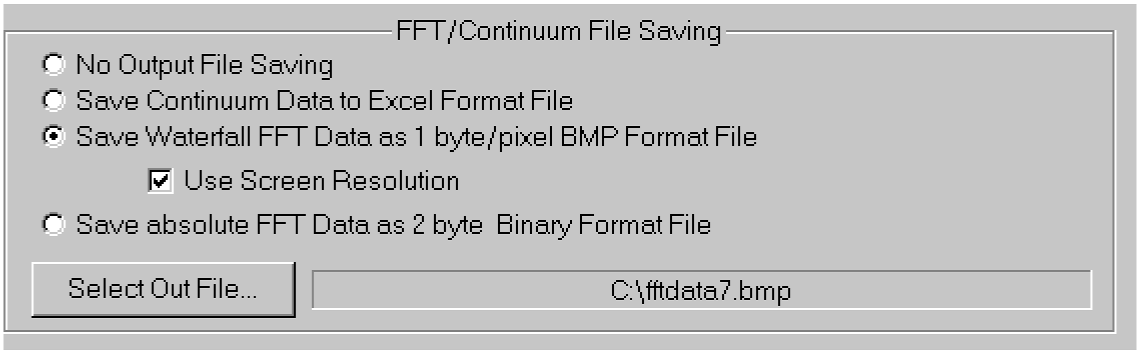
This menu specifies a comma-separated file that will be written with timestamp information, average continuum data and peak continuum data. The following is an example of the data format:

12/02/2003 17:23:01, -108.132, -63.530954

12/02/2003 17:23:06, -108.457, -75.24949

12/02/2003 17:23:11, -108.143, -69.852774

2.3.8.6 Save Waterfall Data as 1 byte/pixel BMP Format File

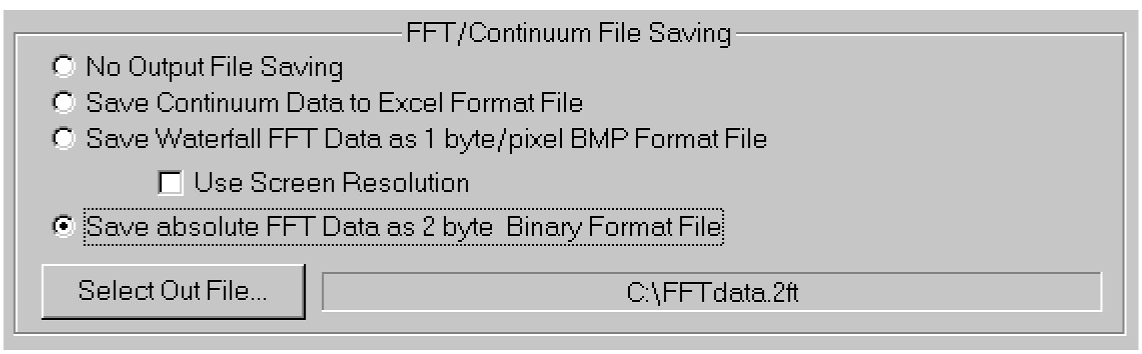


This menu specifies a 1 byte per pixel .BMP graphics file of the waterfall display to be written. If the "Use Screen Resolution box is checked, the image width will be the same as the screen pixel width. This mode creates images that are the same size as the program screen for easy viewing and reduced file size.

If not checked, the image width will be the number of the FFT points within the specified frequency span. This mode can create very wide images depending on the FFT size and span but provides much greater detail than can be obtained on a CRT screen.

The graphics files are limited in size by the Output Size limit. Many graphics programs are limited to how large an image can be displayed as well. Keeping the file size below 100MB is probably a prudent step.

2.3.8.7 Save Absolute FFT Data as 2 byte binary format file

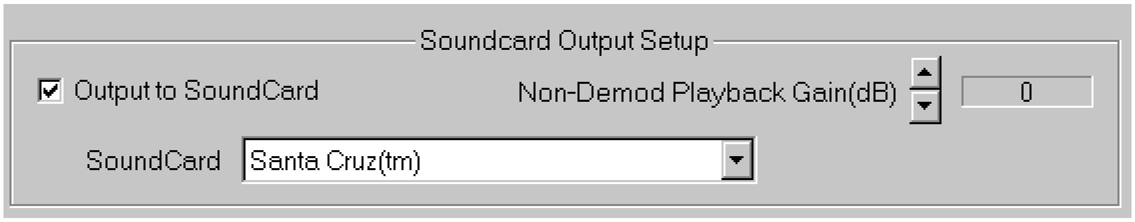


This menu specifies a 2-byte binary format file that will be written with the raw FFT data.

The File format is a 1024 byte header (filled with spaces 0x20 for now) The final format is TBD but will contain key information such as FFT size, sample rates, offsets, etc.

Starting at file location 0x400(1024) the FFT data is placed as a 2 byte signed integer that represents the FFT data times 100. So for example a -135.23dB point would be saved as a 16-bit integer -13523, Little Endean word. All FFT points are saved regardless of the span so the next FFT line would begin at 0x400 + FFTSIZE/2 for real FFTs or 0x400 + FFTSIZE for complex FFTs.

2.3.8.8 Soundcard Output Setup



The Soundcard output menu allows enabling the soundcard for output. This is primarily for listening to demodulated signals but can also be used in playing back wave files or listening directly to I/Q data if the soundcard supports the input sample rate.

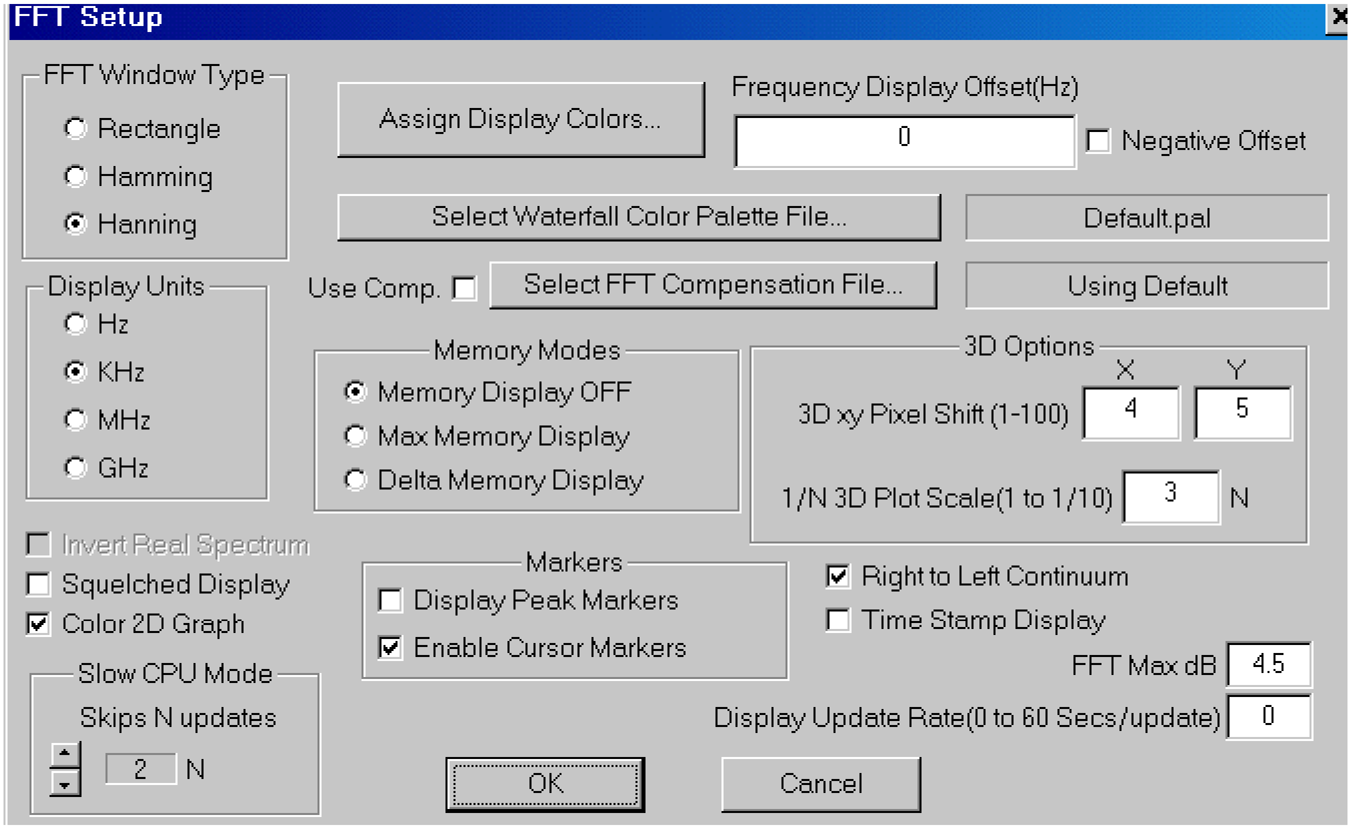
The Soundcard selection menu allows selecting a particular soundcard for multiple soundcard systems.

The "Keep In/Out Rates Same" checkbox will force the soundcard output to run at the same rate as the soundcard input rate. This may be useful for soundcards that do not allow the input sample rate to be different than the output rate. This is only applicable to soundcard demodulation at 48KHz or 96KHz.

The non-demod playback gain control allows the audio to be scaled. This is active only when demodulation is not active, the sound output is selected, and the sample rate is less 100KHz. This is useful for amplifying a captured RF file to a level that can be heard or for retransmission as I/Q data.

2.3.9 FFT Setup

This menu is used to setup a variety of FFT and display options.



2.3.9.1 FFT Window Type

The FFT window type can be selected. The Hanning window is probably the best all around window to use.

2.3.9.2 Display Units

The display units that are used for the display and frequency controls can be specified.

2.3.9.3 Invert Real Spectrum

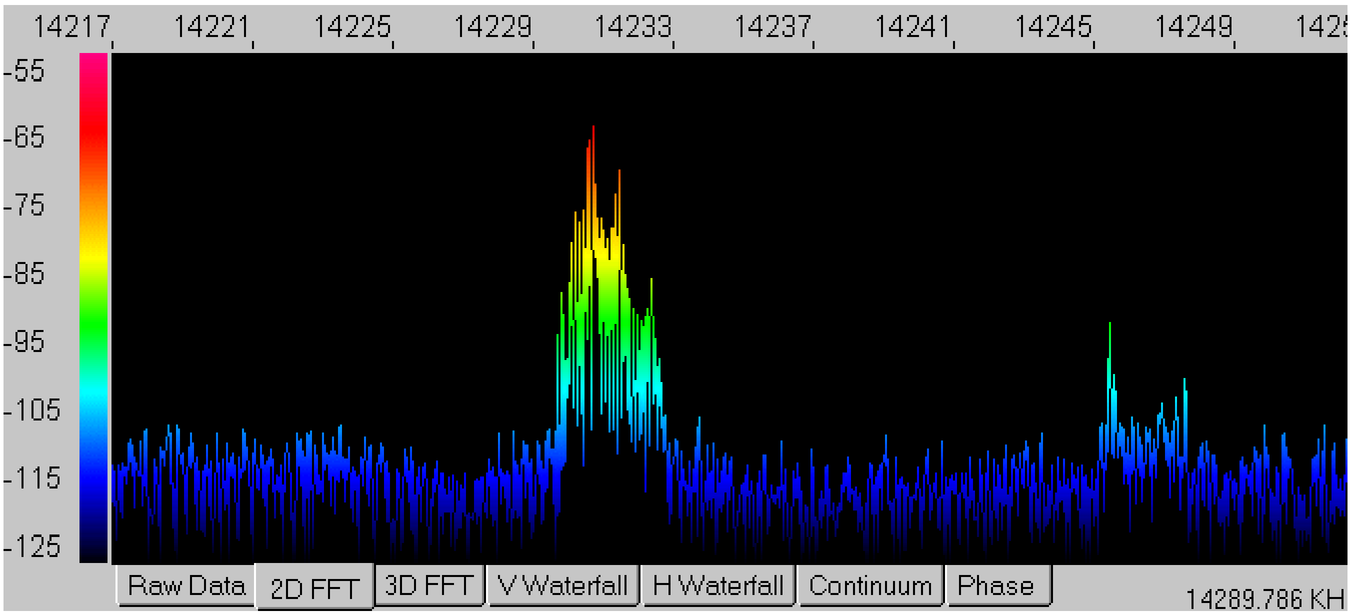
The Invert Real Spectrum box will invert the display frequencies (NOT the actual data) Useful for viewing inverted real spectra.

2.3.9.4 Squelched Display

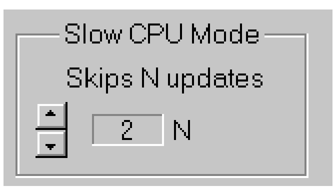
This enables a mode where the display is only updated when the squelch is open when using the demodulation modes. Not only is the display "frozen" when the squelch closes but any FFT data file such as the .bmp capture or binary data file is not written into when the squelch is closed. Any wave file being written into is unaffected by this setting since they have an independent control for qualifying with the squelch control.

2.3.9.5 Color 2D Graph

Selecting this feature changes the 2D FFT display mode so that the graticule is replaced by a graduated color scheme where the 2D display changes color with amplitude using the same palette as the waterfall display.



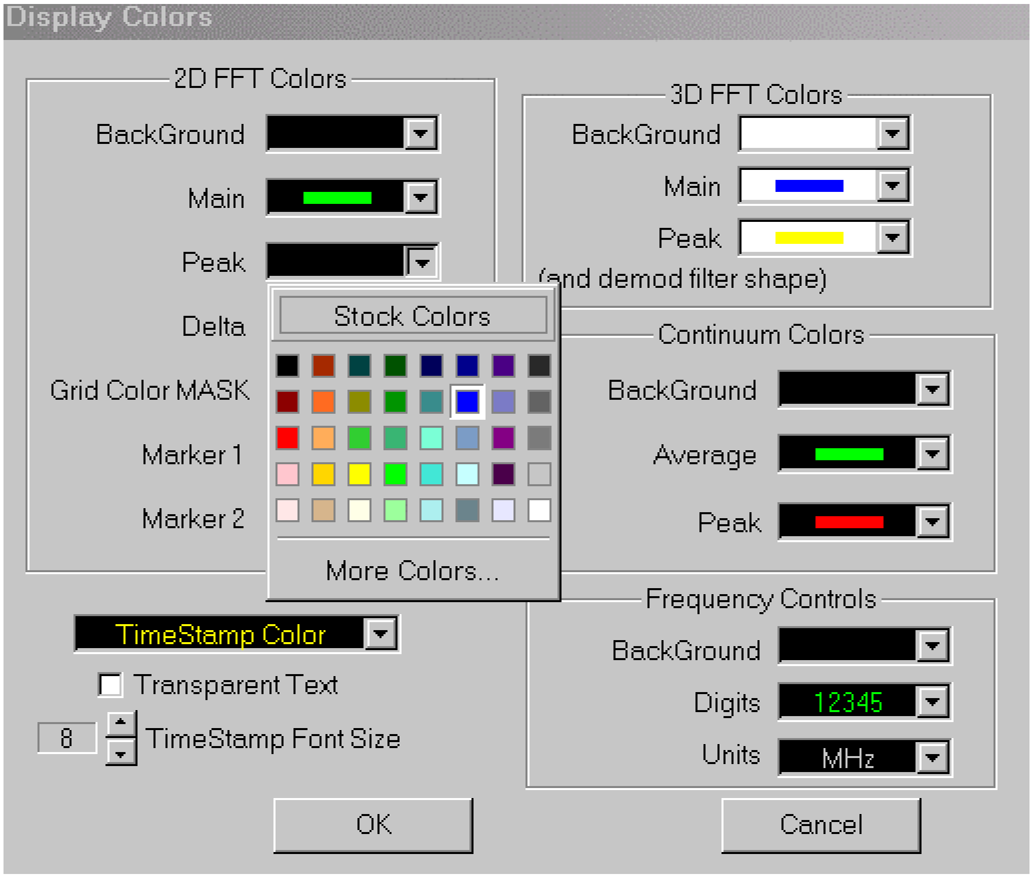
2.3.9.6 Slow CPU Mode



The Slow CPU mode forces the program to skip display updates in order to reduce computational load on the processor. As the number of N grows larger, the more updates that are skipped by the display. A value of zero runs the display at full speed.

2.3.9.7 Assign Display Colors

A submenu allows for changing various display item colors.



2.3.9.8 Frequency Display Offset

The display frequency offset edit box allows the user to specify a frequency offset to all the displayed frequencies. This is useful when external down converters are used ahead of the SDR-14 and the actual input frequency needs to be displayed by the program. The number range is an unsigned 32-bit value (0 to 4GHz). There is a check box to allow this value to be subtracted as well. Keep in mind the center frequency control cannot be negative so there are combinations of offsets that are invalid.

2.3.9.9 Select Waterfall Color Palette File

A submenu allows one to pick a different waterfall palette file for use with the waterfall displays. *.pal files are used that are text files that contain a table with 256 rows representing power (1st row highest power). Each row contains an R,G,B value each with a range of 0-255.

Example format for a couple of rows:

25 255 171

41 232 0

……..

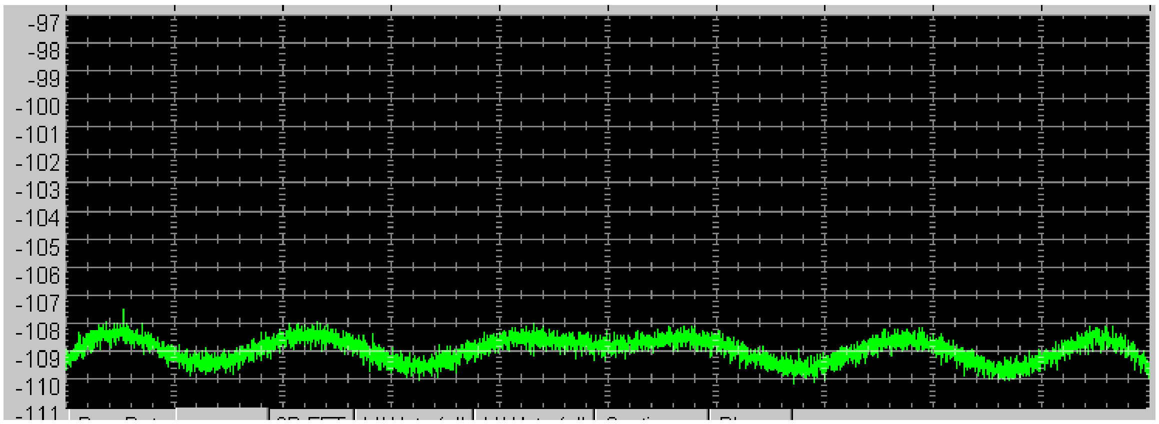
2.3.9.10 Select FFT Compensation File

A submenu allows one to pick a FFT compensation file. A special file can be invoked to flatten out the filter responses of some of the SDR-14 IF filters. This is useful when doing extremely weak signal analysis and removes the ripple from the filter pass band.

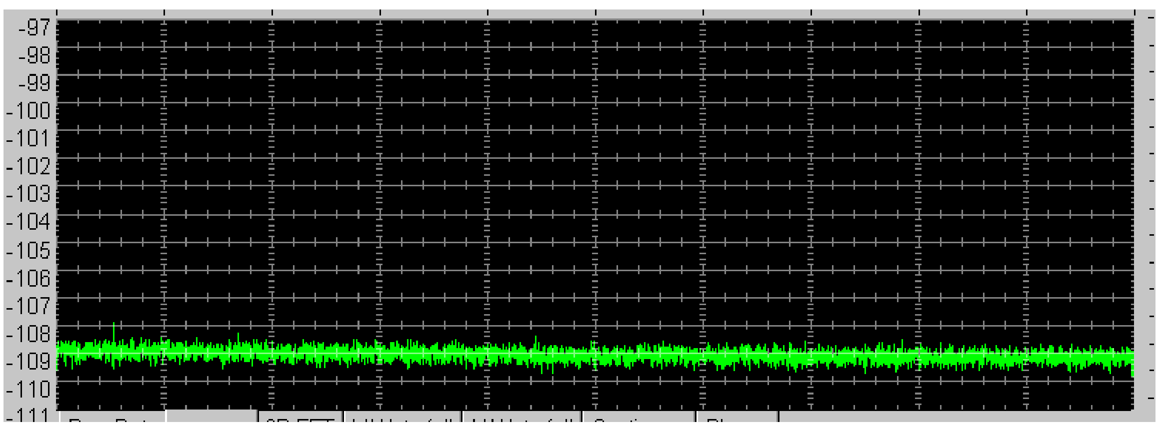
These files have an extension of .fcf and a name corresponding to the AD6620 filter bandwidth that it will compensate.

For example the file "Filter1500.fcf" is used to compensate the 1500KHz BW AD6620 filter setting.

Before:



After:



2.3.9.11 Memory Modes

Three radio buttons are used to select the memory modes.

The max hold mode adds a second trace on the displays that is the maximum value for each point in the display over time. Pressing the "Clr Max FFT" button on the main screen above the "info" box will reset the maximum values.

The delta display mode allows the user to save the current displayed trace into memory and then display a second trace which is the difference between the stored trace and the current FFT trace.

Pressing the "MEM=FFT" button will save the current fft trace to memory.

Pressing the "AUTO delta" button toggles between the delta display (difference between current and stored) and the current and the stored memory traces. All three are displayed on the screen with different colors to distinguish them. The stored memory trace uses the Peak trace color.

2.3.9.12 Markers

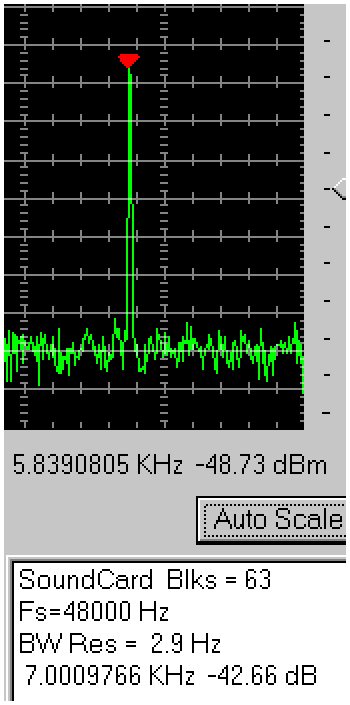
Display Peak Markers - This box enables a peak finding routine that shows the 4 highest power peaks as x's on the display as well as numeric values in the INFO box. This is a more accurate way of obtaining frequency and amplitude since the actual FFT value is used and not the screen accuracy.

Enable Cursor Markers - This box enables the display of the cursor frequency and amplitude just below the display.

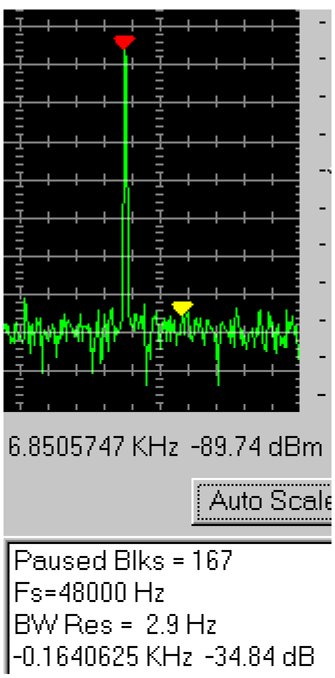
It also enables a display marker system that works in the following way:

Note---Demodulation must be turned off for this measurement mode.

1. Right click on the spectral display near where you want to attach a marker. A red triangle should attach to the FFT display. The F3 to F8 keys can be used to position it exactly where you want. F3, 4,and 5 move it left while F6, 7,and8 move it right at different rates. The numeric value of the marker is shown in the INFO box.



2. Left click on another part of the waveform and the first red marker freezes at the last value of the spectrum and a new yellow marker attaches to the new FFT position. It can also be moved with the function keys. The numeric value of the delta between the two markers is now shown in the INFO box.



3. Right clicking again gets back to step 1. Right click again and the markers are turned off.

2.3.9.13 3D Display Parameters

The 3D xy pixel shift edit boxes allow the user to specify how many screen pixels the 3D display will move every capture. The screen moves left and up.

The 3D scale value allows the user to scale the individual FFT amplitudes within the 3D display.

2.3.9.14 Right to Left Continuum

The Right to Left Continuum allows changing the direction of the continuum scrolling.

2.3.9.15 Time Stamp Display

The Timestamp Display box enables time stamping of the waterfall and continuum displays. Adjust the color, size, and transparency of the timestamp font to give best viewing.

2.3.9.16 FFT Max dB

This edit box allows entering a value to calibrate the display read directly in dBm. This value is added to the FFT values. It typically is around 4dB for the direct input mode.

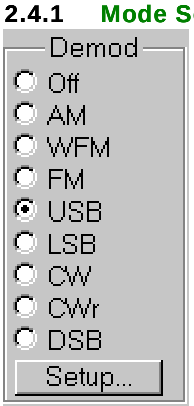
2.3.9.17 Display Update Rate

The Display update rate edit box can be used to cause the display rate to be very slow to allow long waterfall or continuum captures. The value is from 0 to 60 seconds between updates in 1-second steps.

2.4 Demodulator Setup

The demodulation features of SpectraVue are only available when the AD6620 bandwidth is 50, 100, or 150KHz. For Soundcard input it must be around 48KHz.

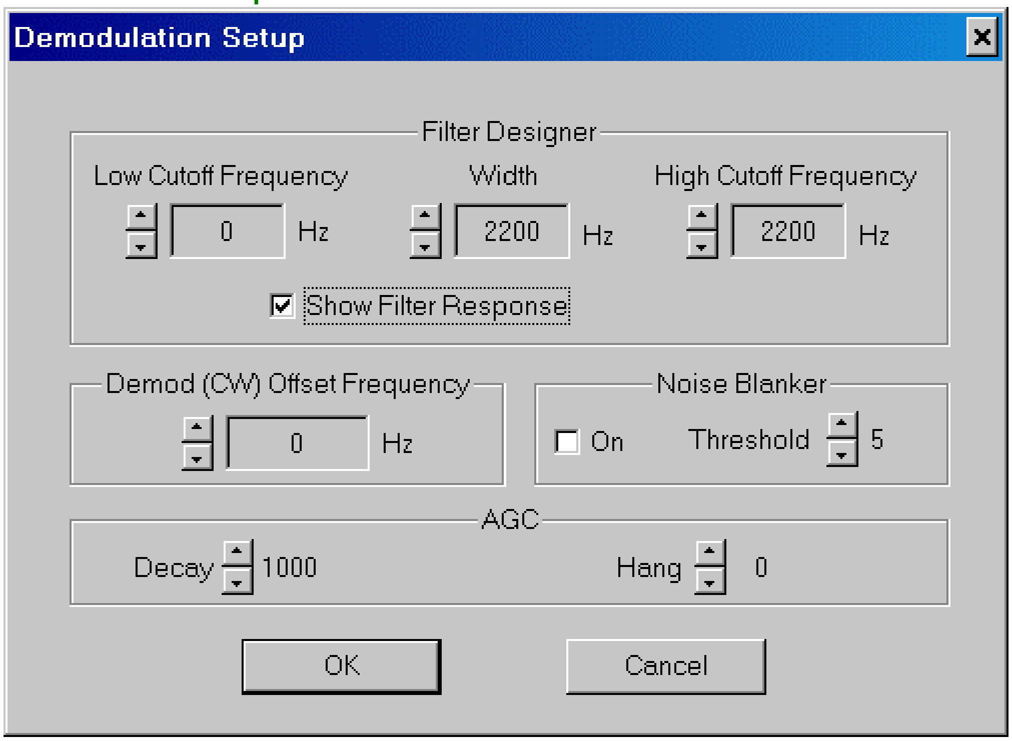
2.4.1 Mode Selection



This menu allows selection of various demodulation modes. Selecting Off will disable the demodulator section and allows selecting AD6620 Bandwidths wider than 150KHz and less than 50KHz.

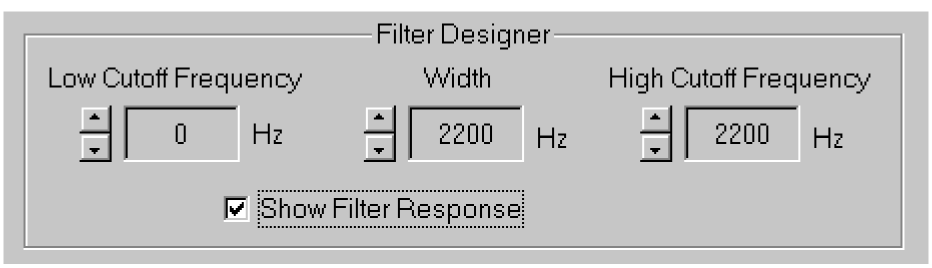
Each modulation mode has its own settings and is modified by pressing the "Setup" button while the desired mode is selected.

2.4.2 Demod Setup



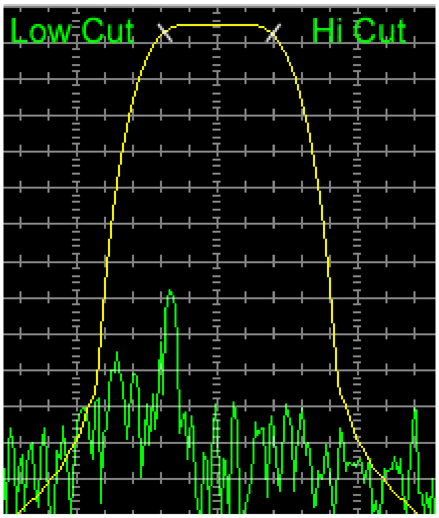
This menu exists for each demodulator mode. Depending on the mode, some items are not selectable.

2.4.2.1 Filter Designer



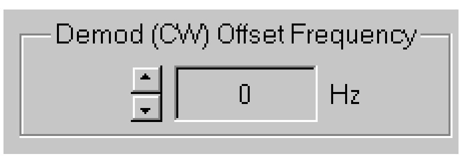
This menu allows the user to customize the demodulator filter width and positions. For single sideband modes (USB, LSB) the low and high cut positions can be adjusted independently. The other modes are symmetrical around the center frequency and so only the width may be adjusted. The filter alpha's are determined by the program and are not user definable.

If the "Show Filter response" box is checked, then the actual filter shape is displayed on the 2D FFT display. The Hi and Low cutoffs are the -6dB points of the filters.



If the high cut off is greater than 3600Hz, then a higher sample rate is used in the filters. This causes a jump in the shape of filters about this transition point. The higher cutoff filters are not as sharp as the narrower ones.

2.4.2.2 Demod (CW) Offset Frequency

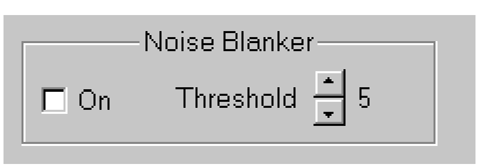


This edit box provides a way to offset the filter and display frequency for receiving CW tones. This is essentially the tone frequency that will be heard in the CW mode when the display frequency is exactly the same as the incoming signal.

This offset can also be used in the USB mode to shift the demodulated signal up in frequency before sending it out. This can be used to output a 12KHz IF signal to the sound card for further decoding by a DRM (Digital Radio Mondiale) decoder. If this offset value is >0 then the demodulator frequency display changes to a center value so the incoming signal is centered as opposed to being at the left side as is normal for USB signals.

For DRM reception, set the offset to 7KHz and the low cut to 0Hz and the high cut to 10KHz. This will center the DRM signal around the 12KHz frequency going out the soundcard or wave file.

2.4.2.3 Noise Blanker



This controls a simple noise blanker algorithm. Checking the On box enables it. Adjust the threshold for best results. Too high a threshold will severely distort the spectrum and signals so use just enough to clip the noise.

2.4.2.4 AGC

This sets the AGC constants. The decay time is in Milliseconds and is the exponential decay time for the AGC. The Attack time is fixed at 10 Milliseconds.

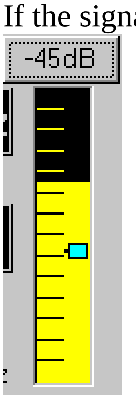
Hang time is in Milliseconds and is a delay time after the signal has gone before the delay time is activated.

2.4.3 Signal/Squelch Bar

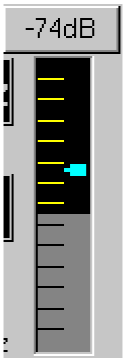
The vertical bar display to the right of the frequency controls is signal strength meter and squelch control.
The height of the bar is the signal strength. If demodulation is off, then the level and height are the peak amplitude within the entire current display span.
If demodulation is active, then the level is the signal strength within the demodulator pass band. The exception is if in NBFM then the level is the FM noise quieting level for use with the squelch.

Clicking on the blue bar within the control or dragging it to the desired level sets the squelch threshold.

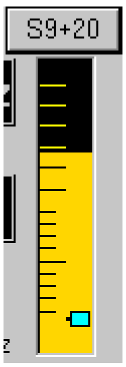
If the signal is above the threshold, the level changes color indicating the squelch is open.



If the signal is below the threshold, it turns grey.



The button above the Squelch control is the peak signal strength within the demodulator filter bandwidth. Pressing the button toggles between dB display and an S meter reading.



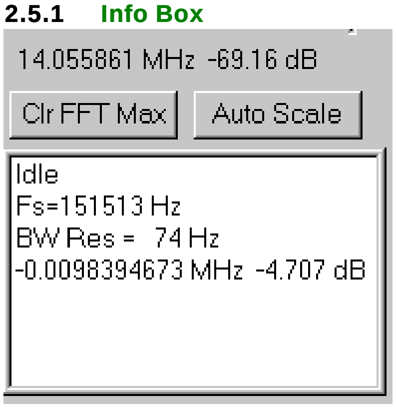
The range is S0 to S9 and S9 +10 to S9 +60dB
The small tics are the S units, the longer ones the "over S9" units.

When in the FM mode, the squelch level is not the signal power but is the output of the FM noise activated squelch so just has a range of 0 to 14.

2.5 Miscellaneous Displays

The program has some other displays scattered around the main screen.

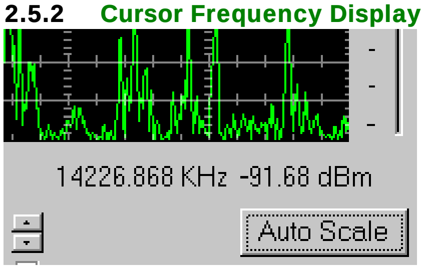
2.5.1 Info Box



This text box displays the program status as well as some information on current sample rates and FFT resolutions.
The first line shows the current status of the program.
The "Fs= xxxxx Hz" is the FFT data sample rate.
The "BW Res" is the bandwidth resolution of the currently selected FFT size and sample rate.

If position markers are active, then the marker information is shown in this box.

2.5.2 Cursor Frequency Display



Just above the info box is the current mouse cursor location if it is enabled and is over a 2D display.

2.5.3 File Progress Bar

The horizontal bar display at the bottom of the screen is active while playing or recording to a file. It indicates the file progress.

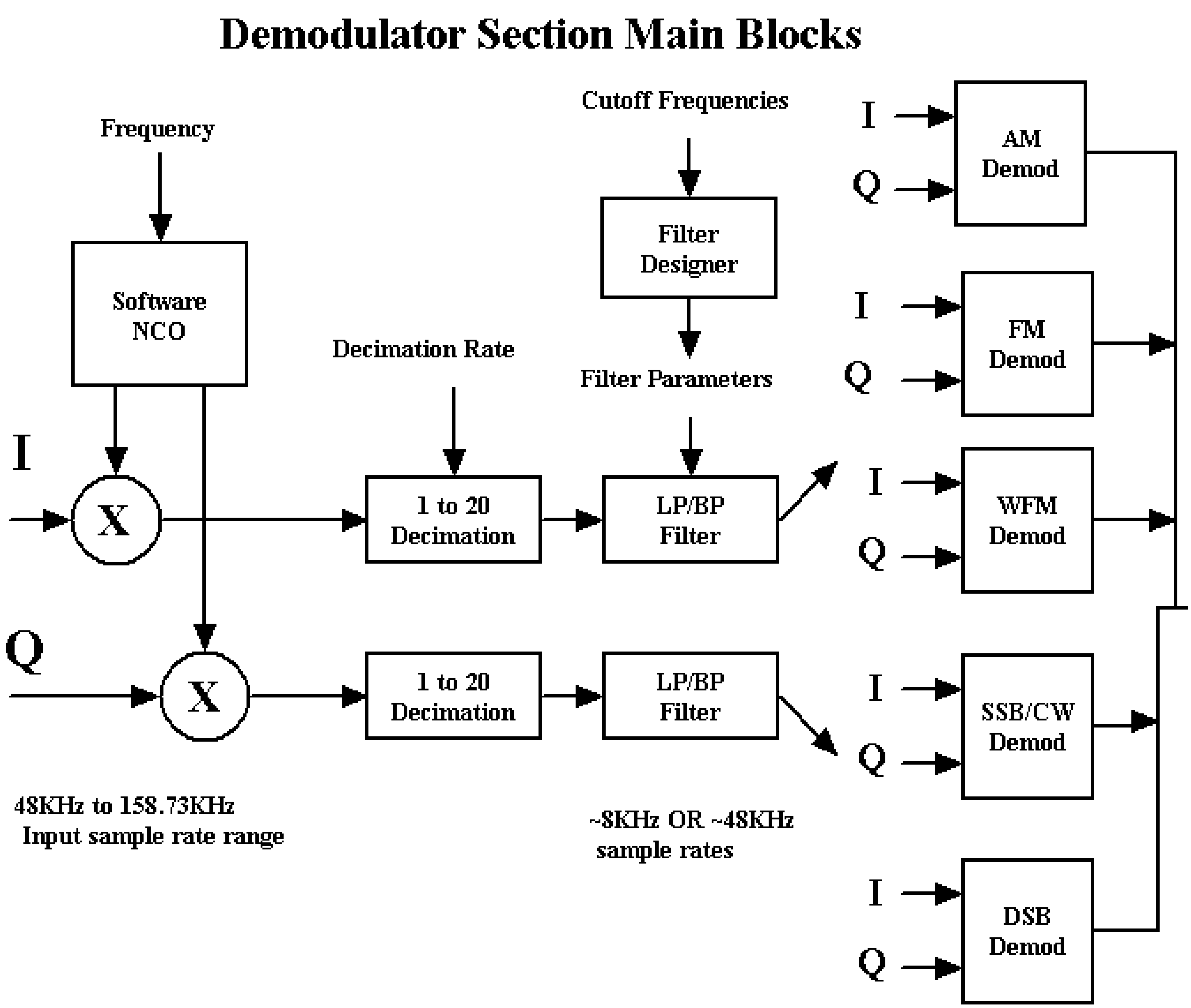
If playing back a file, one can click on the progress bar with the mouse and move to that file playback position. Click on it don't try to drag it across.

3 Technical Details

This Section describes some of the technical details implemented in SpectraVue.

3.1 Demodulator Section

The SpectraVue Demodulator section provides a means to demodulate various signals in real time and output the audio to a wave file or Soundcard. The demodulator is not meant as a high performance receiver but provides a useful addition for actually hearing various signals in real time.



I/Q data from the input source is sent to the demodulator block. It is first "tuned" to the desired signal using a complex mixer and software NCO. This allows one to tune in signals anywhere within the input bandwidth which could be up to 150KHz wide..

Next the I/Q streams are decimated down as close to either 8KHz or 48Khz as possible using integer decimation stages. The decision as to which decimation rate to use is determined by the selected Filter bandwidth. If the bandwidth is greater than 3600Hz then the higher 48KH rate is used.

Next the signal is either low-pass filtered or band-pass filtered depending on the demodulation mode. These filters are designed on the fly depending on the cutoff frequencies specified by the user.

The output of the filters is then presented to the appropriate demodulator block that takes the I/Q signal and extracts the specified modulation as a real audio data stream.

Finally the real audio stream is re-sampled so that the output sample rate is exactly 8KHz or 48KHz. The soundcard runs at a fixed 48KHz while the wave file data can be either 8KHz or 48KHz.

3.1.1 NCO/Mixer/Decimation Stages

This stage shifts the desired signal frequency to zero for base band processing.

The decimation stages down sample the input from 1 to 20 to get as close to 8 or 48KHz as possible before entering the main filter routines.

The Wideband FM demodulator is a special case where the FM is demodulated before the decimation stages in order to use the full input bandwidth.

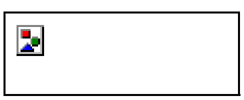
Below is a C code excerpt of the complex mixer/NCO and first decimation stage.

```
for( i = 0; i<length; i+=2 ) // put new samples into Queue
{
     offset.x = cos(m_Timeinc);     //Create NCO values
     offset.y = sin(m_Timeinc);
     m_Timeinc += m_SoftNCOInc;     //update NCO
     acc.x = m_DCoffset+(double)(pIn[i]);  //get I/Q sample
     acc.y = m_DCoffset+(double)(pIn[i+1]); //add DC offset
//Complex multiply input by NCO data
     m_pCpxQue[m_State].x = ((offset.x*acc.x) - (offset.y*acc.y));
     m_pCpxQue[m_State].y = ((offset.x*acc.y) + (offset.y*acc.x));
//decimate by m_Val filter
     if( ( (++m_SampCnt)%m_Val ) == 0 )//calc first decimation filter every m_Val samples
     {
          acc.x = 0.0;
          acc.y = 0.0;
          Firptr = m_pCpxQue;
          Kptr = m_pCoef + m_FirLen - m_State;
          for(j=0; j<     m_FirLen; j++)     //do the MAC's
          {
               acc.x += ( (Firptr->x)*(*Kptr) );
               acc.y += ( (Firptr++->y)*(*Kptr++) );
          }
          pOut[n++] = acc;
     }
     if( --m_State < 0)     //deal with FIR pointer wraparound
          m_State = m_FirLen-1;
}
```

3.1.2 Demodulation Filters

The demodulation FIR filter is calculated in real time from the user input cutoff frequencies.

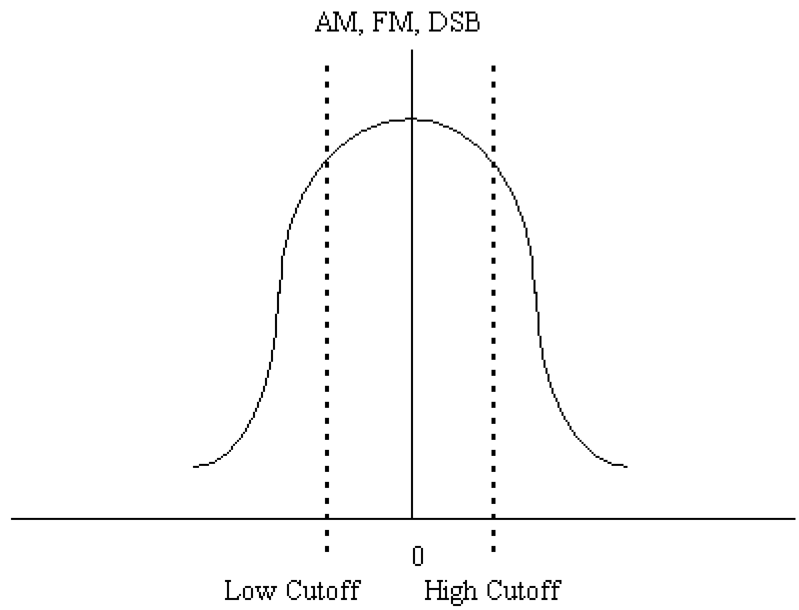
An ideal LP FIR filter is first designed:

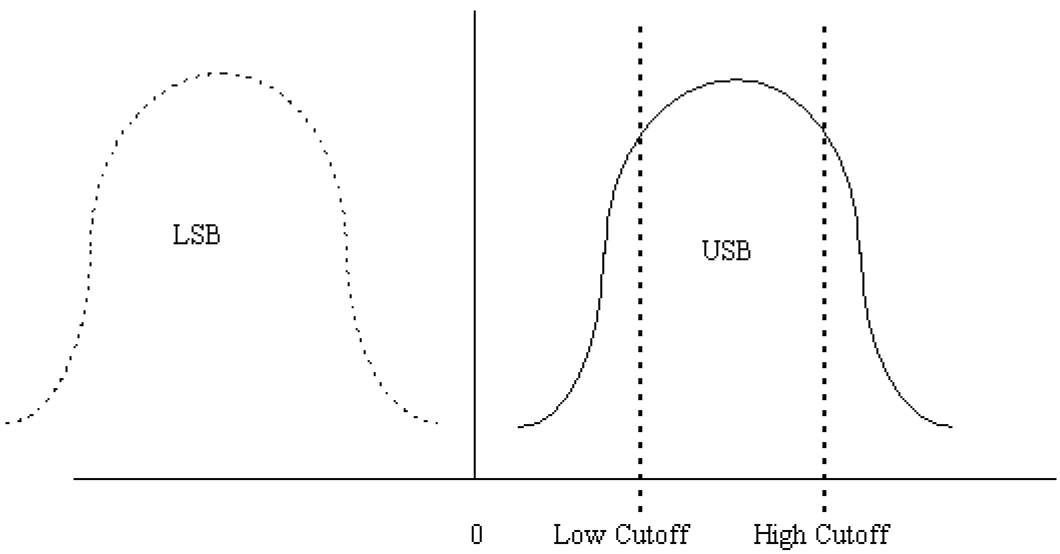


The coefficients are then windowed with a Kaiser Window to reduce the side lobes. The maximum FIR size is limited to 250 taps.

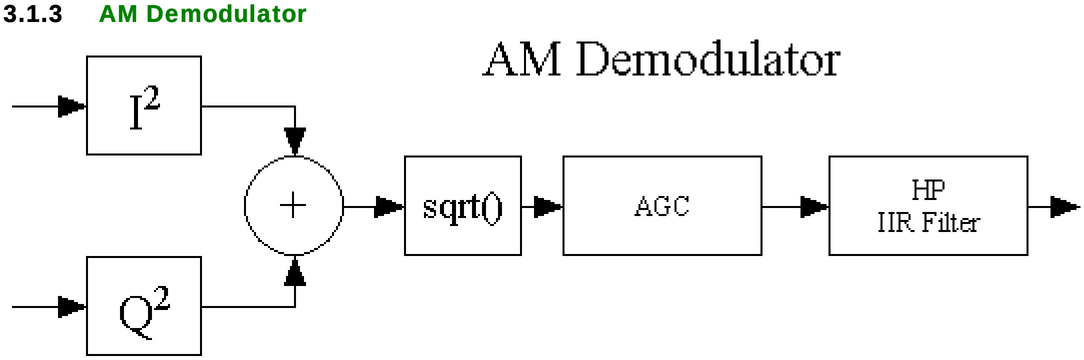
The demodulator uses two filter types. The AM.FM, and DSB demodulators require a LP filter on I and Q since these signals are symmetric around the zero frequency.

The SSB and CW filters use a BP filter on I and a matching BP filter on Q with a Hilbert 90 degree phase shifter. The BP filters are designed as LP filters that are then shifted to the proper cutoff frequencies and the Q side filter coefficients are further processed to introduce a 90-degree phase shift.





3.1.3 AM Demodulator



The AM demodulator takes the I and Q data and converts it into magnitude data.

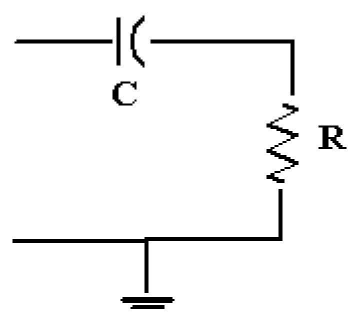
Mag = sqrt(I*I + Q*Q)

The magnitude is fed to an AGC block to provide some gain control. Then a simple high pass filter is used to remove the DC component of the magnitude leaving the audio.

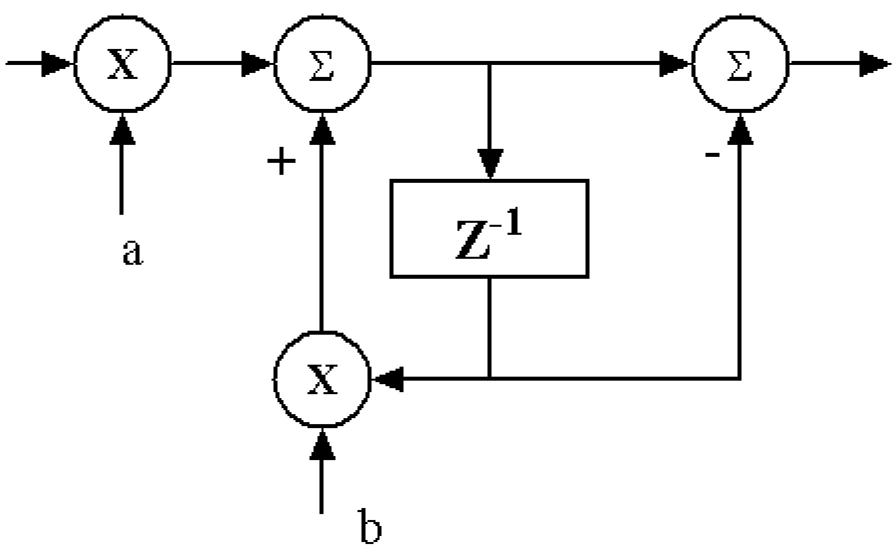
The following is the digital equivalent to a simple RC high pass DC blocking filter.

$$H_z = \frac{a[1 - z^{-1}]}{[1 - bz^{-1}]} \quad \text{where } a = \frac{1}{(1 + \frac{T}{2RC})} \quad \text{and} \quad b = a(1 - \frac{T}{2RC})$$

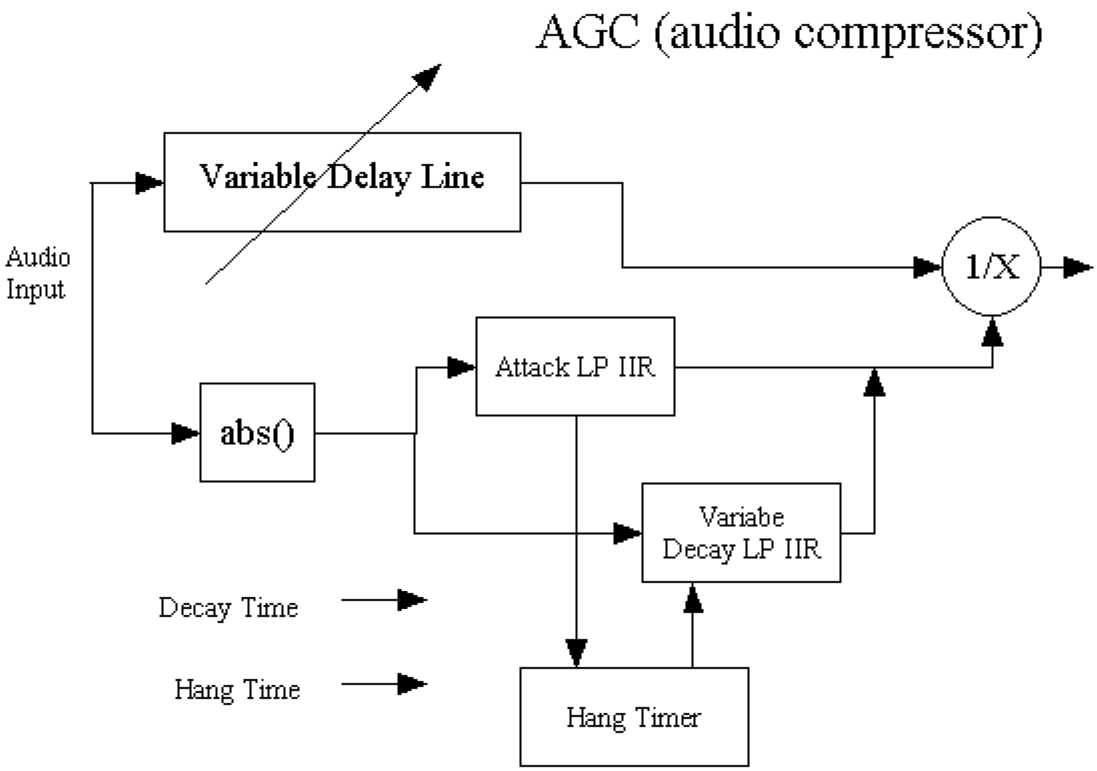
ANALOG HP



DIGITAL HP



3.1.4 AGC Compressor

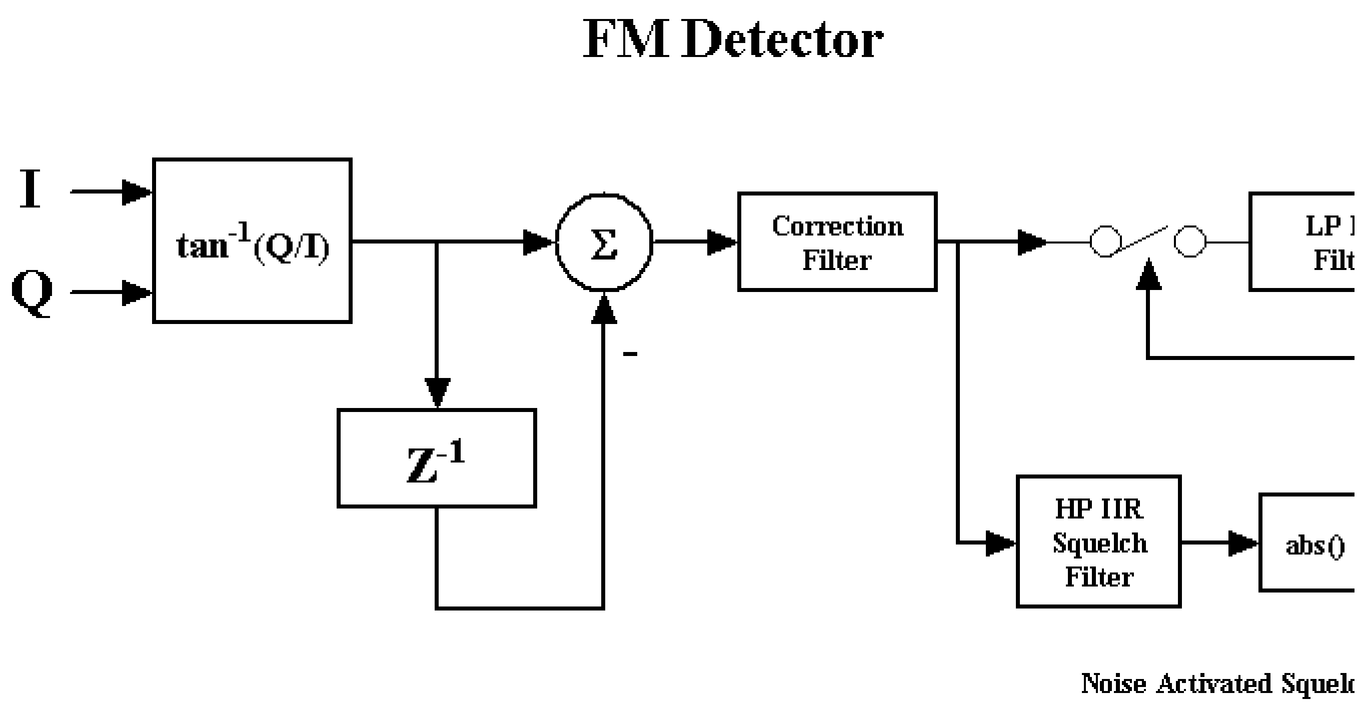


The demodulator AGC uses a simple audio compressor scheme with variable decay filters and also a hang timer.

The Variable Delay Line allows the AGC to "look ahead" in time and have the new gain value ready before the leading edge of a signal is processed.

3.1.5 FM Demodulator

The FM demodulator follows the following block diagram.

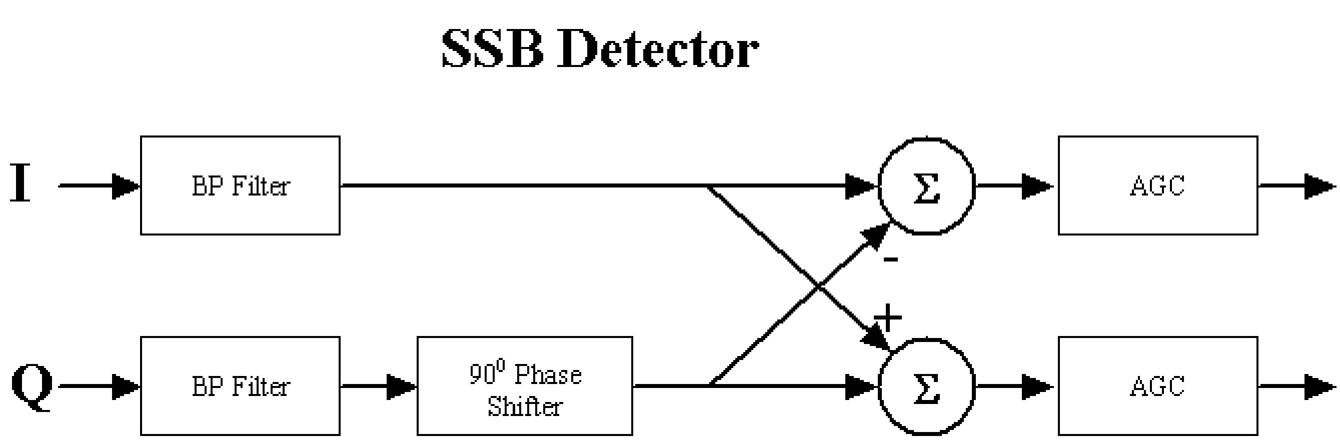


The actual implementation combines the arctan function and differentiation and correction filter in one 6 tap filter as described in the book by Marvin Frerking, "Digital Signal Processing in Communications Sytstems", page253-256.

A noise gated squelch is implemented by high pass filtering the output of the detector, taking its magnitude and then low pass filtering the magnitude. As an FM signal increases in strength, the high frequency noise in the pass-band above the normal audio range decreases rapidly giving a better means to gate a squelch circuit than just using the signal magnitude.

3.1.6 SSB/CW Demodulator

The SSB/CW demodulator follows the following block diagram.



The SSB/CW demodulator uses two identical band pass filters which provide the final demodulator band width. The Q data is shifted by 90 degrees and then added or subtracted from the I data to obtain the upper or lower sideband audio.

The BP filters are designed as Low pass filters and then converted to tow band pass filters using the following method:

a. Design a lowpass FIR filter with a passband of ½ the desired BP passband and also to whatever stopband goals.
b. Convert the LP coefficients to I and Q BP FIR coefficients using the following equations.

$$h_{IBP}(n) = 2h_{LP}(n)\cos(2\pi f_0[n - \frac{(N-1)}{2}]T)$$

$$h_{QBP}(n) = 2h_{LP}(n)\sin(2\pi f_0[n - \frac{(N-1)}{2}]T)$$

where:

$h_{LP}(n) = \mathrm{n^{th}}$ Lowpass FIR Coefficient

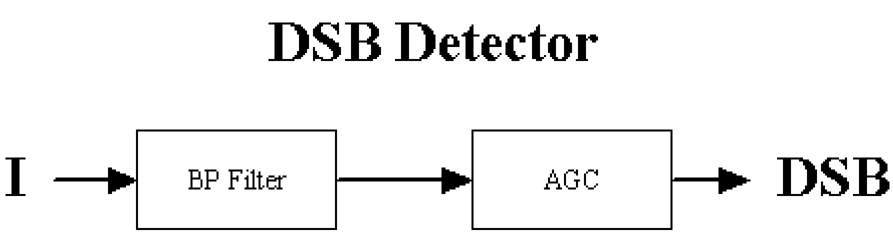
f_0 = Bandpass Center Frequency

N = Number of Coefficients

T = Sample Period

3.1.7 DSB Demodulator

The DSB demodulator follows the following block diagram.



The Double sideband detector just uses the I data and band pass filters it.

3.1.8 Re-sampler

The SDR-14 sends data to SpectraVue at various data rates depending on the bandwidth chosen and actual clock rates. These rates are typically not an integer multiple of standard soundcard rates so a means to convert by a fraction was needed.

Resampling ratio = L/M

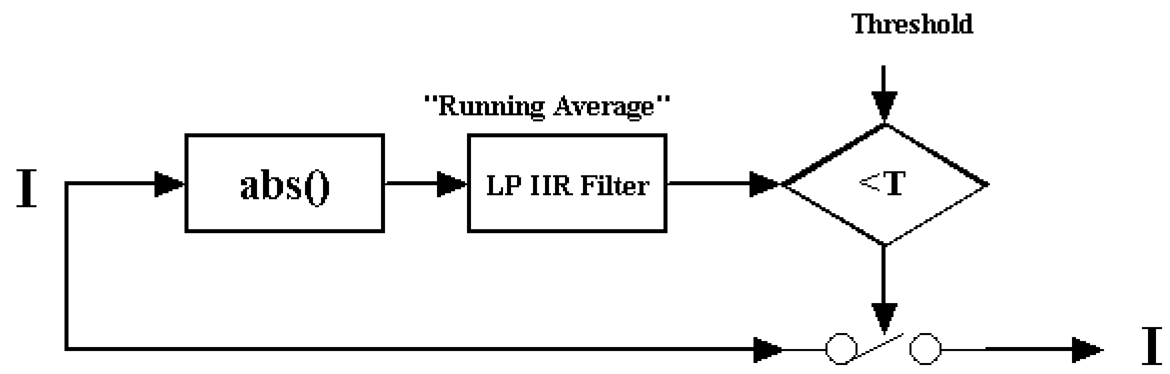
The basic idea is to perform an integer interpolation up by a large number, then decimate back down to the desired rate using a second integer value. By using a few "tricks" one can reduce the filtering overhead and provide good aliasing performance without using up too much CPU time.

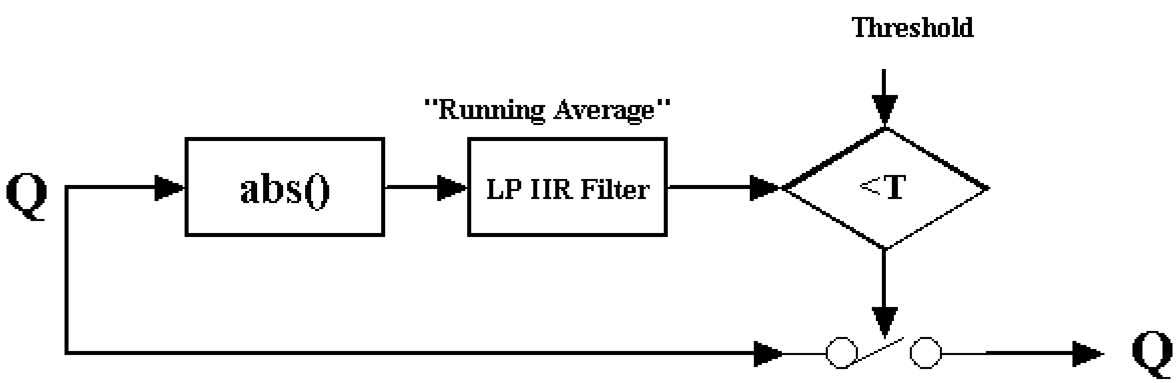
The following is a code segment of the main re-sampler loop.

```
for( int i = 0; i < InLength; i++ )
{
    while( phase < (double)m_InterpRatio )
    {
        int h = 0;
        int k = (int)phase;
        phase += m_DecRatio;
        int c = m_CoefLength - 1;
        double* pCoef = m_pCoef[k];
        double output = m_pHistory[h++] * pCoef[c--];
        for( int ii = 2; ii < m_CoefLength; ii++ )
            output += (m_pHistory[h++] * pCoef[c--]);
        output += (pIn[i] * pCoef[0]);
        pOut[j++] = output;
    }
    phase -= (double)m_InterpRatio;
    // Update history array
    for( int m = 0; m < m_CoefLength - 1; m++ )
        m_pHistory[m] = m_pHistory[m+1];
    // Last input
    m_pHistory[m] = pIn[i];
}
```

3.1.9 Noise Blanker

The noise blanker function operates on the I/Q signal prior to any filtering by the demodulator. It is a very simple minded algorithm which looks at small sections of the waveform at a time and keeps a running average of the absolute magnitude over that section. If the main signal ever goes above this average by a specified threshold value, then the main waveform is set to zero until the signal drops below the threshold. This basically finds any pulse type signal and zeros it out.





3.2 FFT Implementation

The primary FFT algorithm used by SpectraVue is a modified version written by Takuya Ooura. (http://momonga.t.u-tokyo.ac.jp/~ooura/)

It uses his radix 4 FFT package and has been modified to provide a lot of post processing such as averaging, conversion to decibel power, peak hold, etc.

The FFT can be looked at as N number of band-pass filters each looking at a different frequency of the input frequency. The FFT is just an algorithm that efficiently implements this large number of filter banks. Each frequency "bin" is evenly spaced across the sampling frequency range. For real data inputs the range is only half the sample rate. The bin band width is the sampling frequency divided by the number of FFT points. SpectraVue implements FFTs from 2048 points to 262144 points.

The amplitude of the FFT is converted to power in decibels by taking the Log of the FFT output. It is scaled so that the minimum power level is -140dB and the maximum value is 0dB. This maximum value is assumed to be the largest 16 bit sin wave input possible. Since the output of the SDR-14 and soundcards is 16bits, the input signal can never be greater than this.

A pure input sin wave ... Asin(wt)... will produce an fft output

peak of (N*A/Kx)^2 where N is FFT_SIZE.

Kx = 2 for complex, 4 for real FFT

To convert to a Power dB range:

PdBmax = 10*log10((N*A/Kx)^2 + K_C) + K_B

PdBmin = 10*log10(0 + K_C) + K_B

if (N*A/Kx)^2 >> K_C

Then K_B = PdBmax - 20*log10(N*A/Kx)

K_C = 10 ^ ((PdBmin-K_B)/10)

4 Operation Hints

4.1 Calibration

The following ideas are presented to help one calibrate the SDR-14 and SpectraVue for applications where wither frequency or power accuracy is needed.

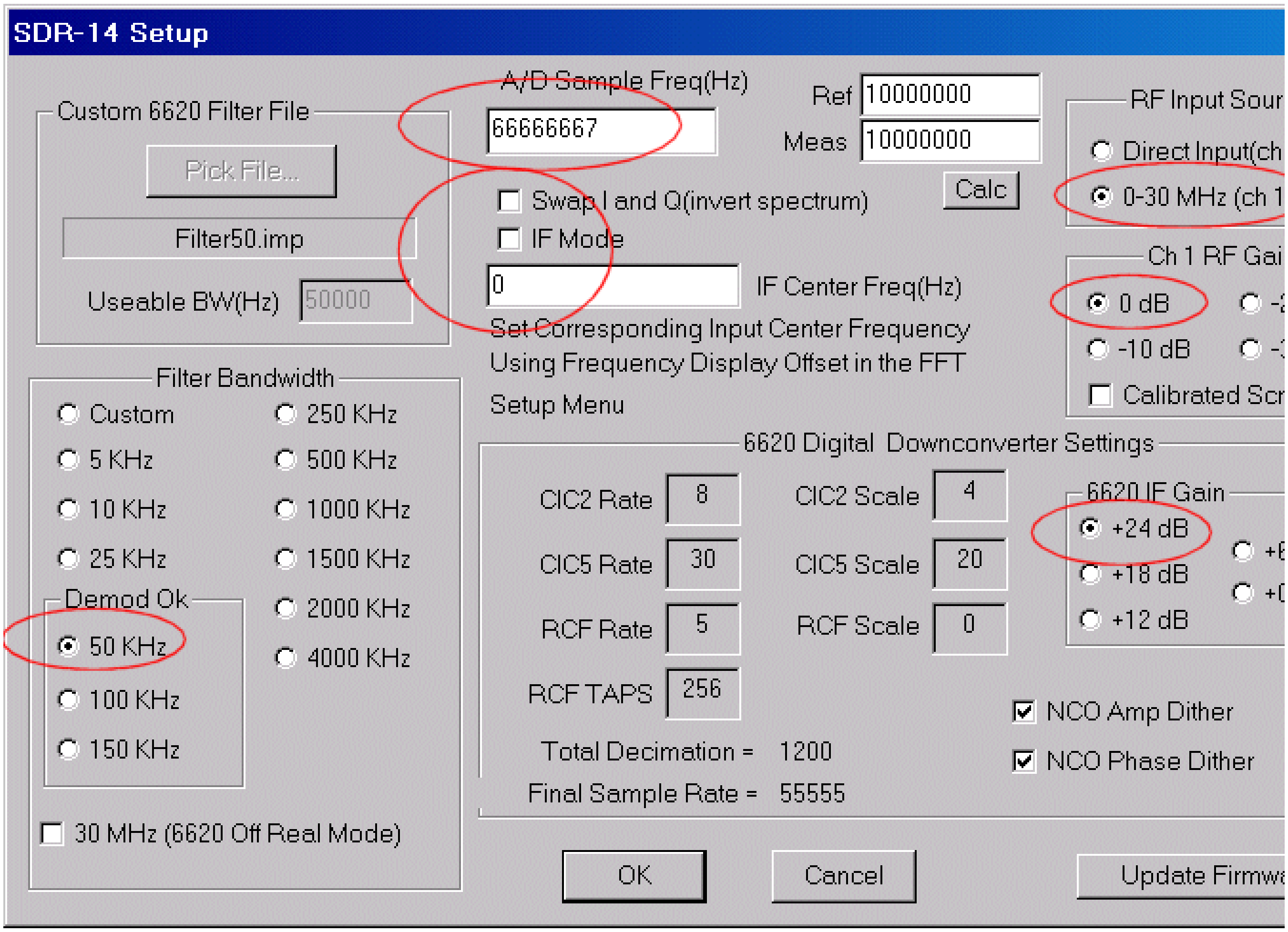
4.1.1 Sample Clock Frequency Calibration

The internal sample clock of the SDR-14 is a crystal oscillator running at a nominal frequency of 66.666…Mhz. It is not oven controlled or adjustable so some means is needed to correct for it in SpectraVue.

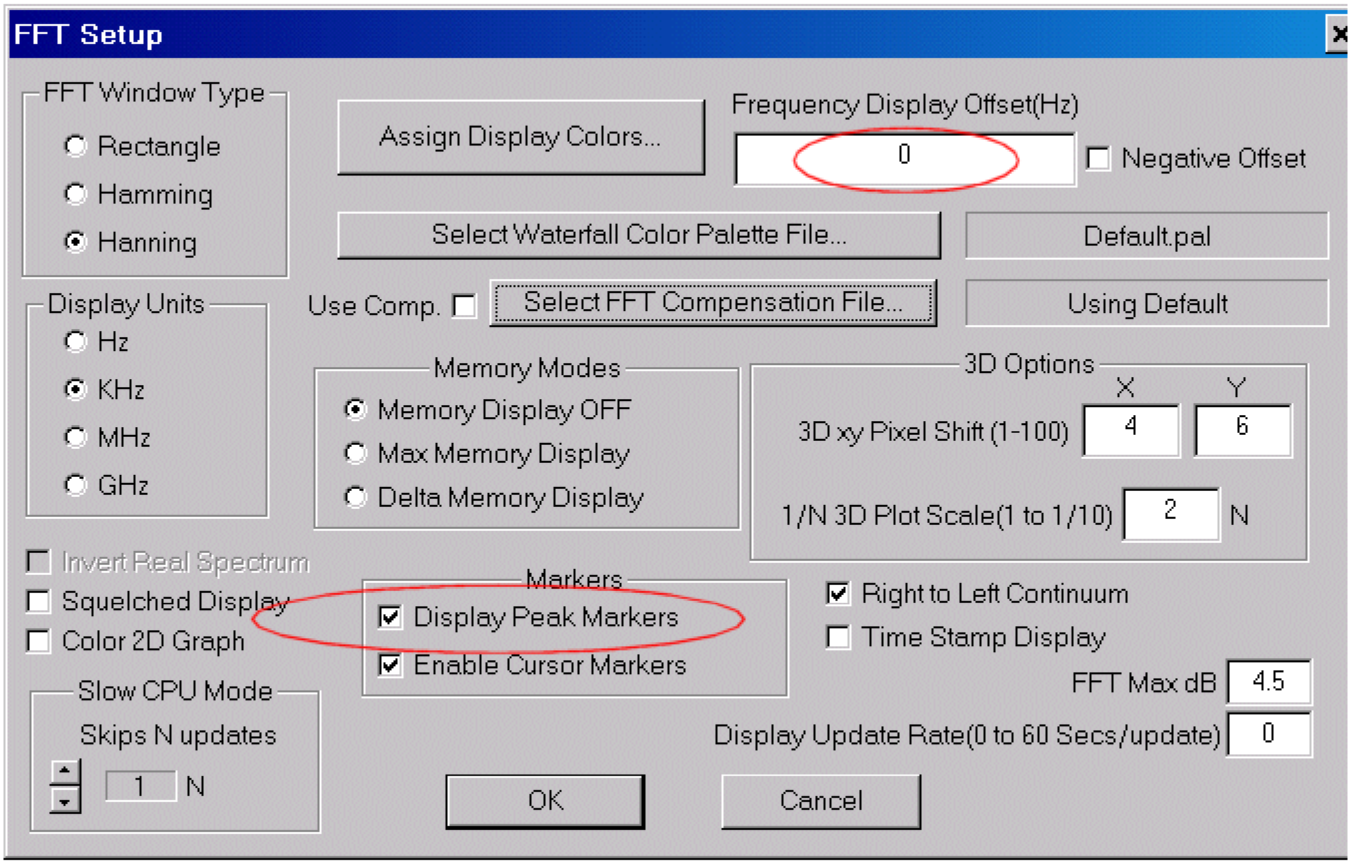
The easiest way to calibrate is to input a known reference frequency into the SDR14 and accurately measure it using SpectraVue. The difference in the measured and actual frequency can then be used to change the value SpectraVue uses for the sample clock. WWV can be used if a decent signal is available or a good lab source. A local AM station carrier could also be used to get reasonably close.

The following SpectraVue setup can be used to help calibrate the system. After it is up working, one can save this calibration setup to the .ini setup file to make life easier next time you want to calibrate.

1. Setup SDR-14 for a 50Khz BW, and set the A/D Sample Frequency to nominal of 66666667Hz if not already there. It doesn't have to be but for this example it is what it would be the first calibration time.

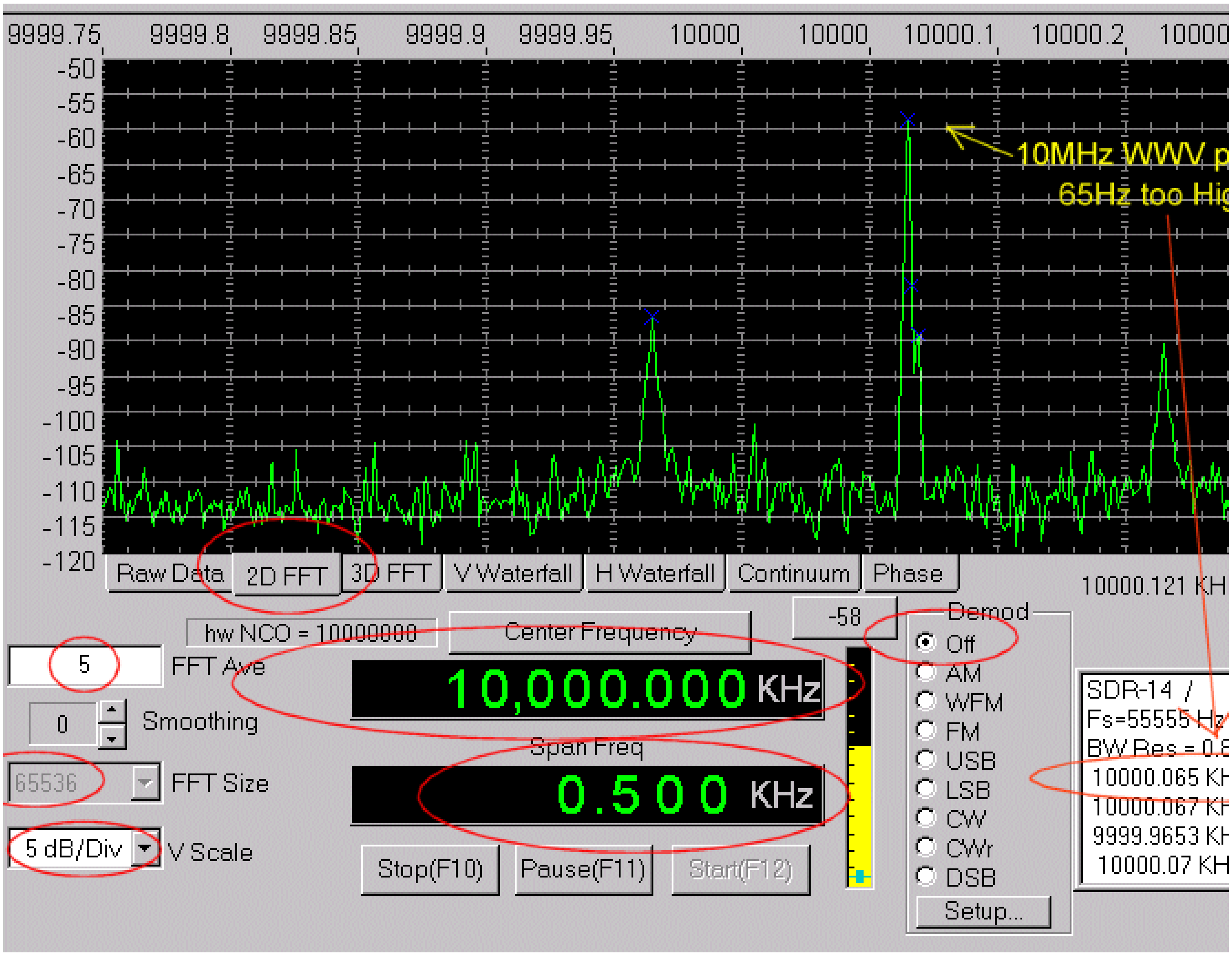


2. Set the FFT setup menu as shown with the "Display Peak Markers" button checked.

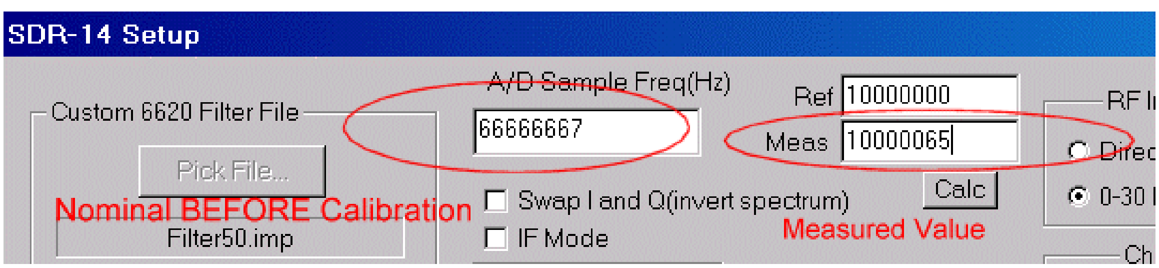


3. Setup the main screen as follows with an FFT size of 65536, an average of 5 or so, a center frequency of 10MHz(for WWV), demod OFF, and a span of 500Hz.. Press Start and adjust the screen to allow easy viewing of the 10MHz peak. Note the little blue x on the peak and its corresponding marker frequency of 10000.065Khz in the info box. This is the measured frequency of WWV and shows that the program is off by 65Hz.

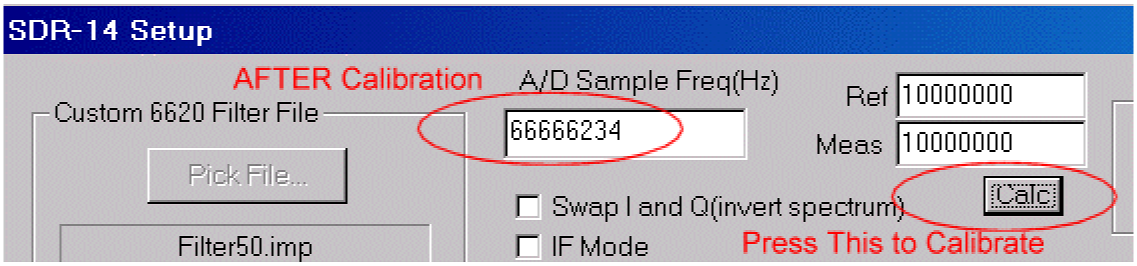
Remember the program can only measure frequency as accurately as the FFT BW resolution so make sure you use a large enough FFT to get within 1 Hz or so. The example show a .85HzBW so is ok.



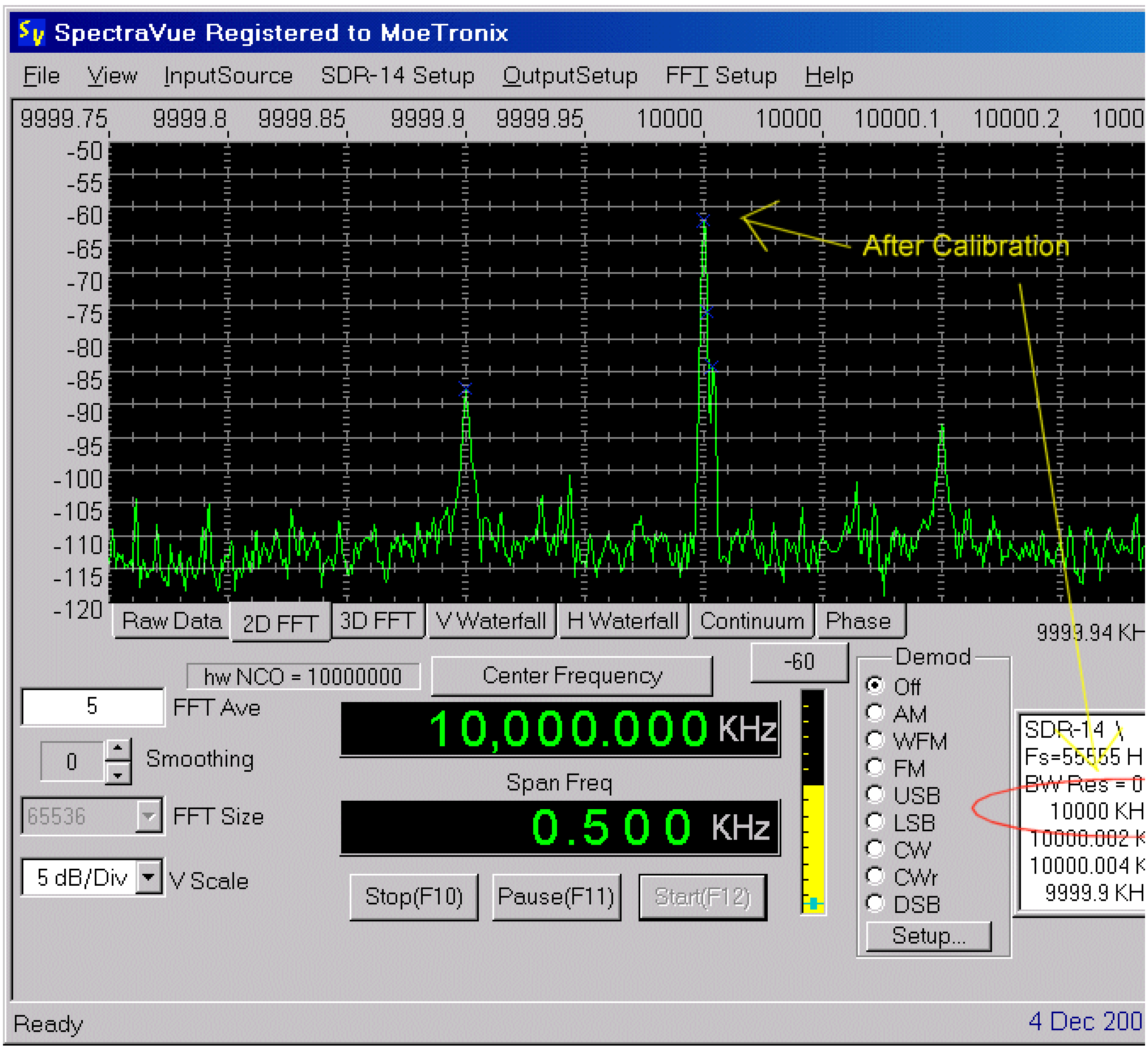
4. Stop the capturing and open the SDR-14 Setup menu. Notice the measured frequency has been automatically placed in the box(if you had peak markers checked and it was marking the correct reference peak) One can manually enter the Reference and Meas frequencies if not using a 10MHz reference.



5.Click on the "Calc" button and the new A/D Sample Frequency is calculated. The "Meas" box will revert to the Ref frequency so one doesn't accidentally keep scaling the A/D sample frequency.



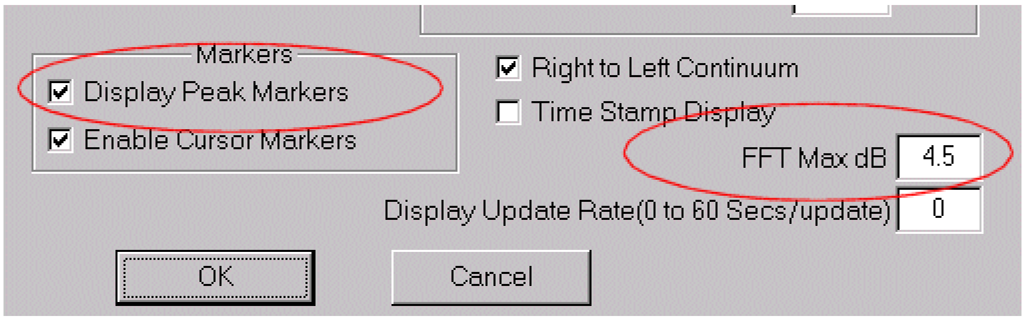
6. Go back to the main screen and start the captures again to verify that the new calibration has taken affect. One may have to perform this a few times to iterate the error to zero if the signal is noisy or the SDR-14 is still warming up.



4.1.2 Amplitude Calibration

The easiest way to calibrate the SDR-14 and SpectraVue for power amplitude is to simply input a sine wave signal of known power and adjust the "FFT Max dB" value in the FFT setup menu. This number is simply added to all the screen values.

For example, set ones signal generator to say -20dBm and measure the amplitude shown on the 2D FFT screen. Use the Peak Markers for best accuracy rather than the cursor marker since the screen resolution is not very accurate. Enter a value into the "FFT Max dB" edit box in the FFT setup menu that when added to the measured value will make it display -20dBm. The value can be positive or negative.



In the direct input mode, a value of 4.5dB will be very close since this is the level in dBm where the A/D begins to clip. If the internal preamp or an external one is used, then one must change the value accordingly.

4.2 Application Examples

As various applications are written up, they will be placed in this section.