

Modulation – Demodulation Software Radio

Yahoo user group: <https://groups.yahoo.com/neo/groups/mdsradio/info>

MDSR website: <http://users.skynet.be/myspace/mdsr>

Build your own IF SDR Introduction of MDSR V3.0

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Accomplishments since the last publication

Base Station setup

The MDSR test station currently in use is a modified FT-950 from Yaesu. The transceiver is connected to a Hy-Gain 18HTjr vertical antenna on top of a 3-storey building. Ground is connected to an iron riser pipe. The station is located on a foothill 10 miles north of Vancouver. Power output is 20W for digital modes and 100W for voice.

The computer running the MDSR software is an i5 processor running at 3.4GHz with 8GB of RAM. The installed operating system is either 64bit Win 8.1 or Ubuntu 14. The sound card installed is an ASUS Xonar Essence true 24bit Performance. The on-board sound device is a Realtek High Definition on-board audio device.



Mobile setup



The mobile setup is a modified IC-7000. The antenna is a telescoping HF antenna from Hy-Gain (MFJ-1979) with homebrew loading coils for 40 and 80m (left); used for stationary operations. A 4-band antenna (CR-8900 - 10m, 6m, 2m, 70cm) is used while driving (right). The home made antenna rack fits multiple vehicles and can be transferred from one vehicle to the other in minutes. The rack includes grounding rods that can be connected on each end for better performance in stationary operation.



Mobile setup of MDSR in a Honda Element



The IC-7000 radio is mounted in the lockable glove compartment which is closed while the car is parked or the radio is not in use (computer and BiLIF are packed away in a box).

The silver box on the dash contains the BiLIF unit and the Xonar U7 USB soundcard, made by ASUS.

The computer running the MDSR software is an ASUS Transformer T100 book running Win 8.1 32bit. It also runs the digital communication programs for JT-65 and WSPR which use the internal sound card.

A USB hub connects all the devices together. The IF and the PTT connection come from the transceiver and plug into the BiLIF.

Note: The IC-7000 does not need a control head – all functions are performed by the MDSR software via CAT control. The IC-7000 control head is only used during mobile operations.

Portable Setup

The portable setup consists out of a modified FT-817, a silver box that contains the BiLIF and the USB sound device. The mic/headset plug connects to the web-book internal sound card for digital operation.



The Portable MDSR system packed away in 2 travel cases for easy transportation

On the left (smaller case) the antenna utility box contains all various antenna parts, such as loading coils, antenna extenders and the mounting bracket. The long rod across both cases is the MFJ-1979 telescopic antenna which extends to 5m (15ft.). It does not fit into the cases and has to be carried separately.

In front of the small case is the Asus T-100 Transformer book, which runs the MDSR software.



The big case contains all the wire connectors, chargers, power adaptors, hub, spare batteries (8 x 2500mA/h NiMH) and the antenna cable (hidden behind the lid cover) and the BiLIF-U7 sound device box (silver). The ASUS T-100 can be wrapped in towels and laid on top of the FT-817 for transportation. The standard NiCad battery pack in the FT-817 has been replaced with 8 2500mAh NiMH rechargeables. The battery pack has been modified to allow charging of the batteries without removing them from the radio.

The whole setup weighs less than 15lb. One charge lasts for about 4h of operation (TX output reduced to 2.5W). The batteries can either be charged from the mains or with 12V from a cigarette lighter adaptor.

Awards received and in progress

- WAS50 in mixed bands for JT-65 mode
- Worked all continents of the world in mixed bands for JT-65 mode
- 100 Grid Squares in mixed bands for JT-65 mode
- Worked 53 states (confirmed 44 on e-qsl and 31 on QRZ.com) as of June 2015

Hardware

The LIF2014 kit was released. The schematics and the components are identical to the previous LIF2012 kit that has worked so reliably. The only update was the footprint of L1 to make it easier to assemble the unit.

A lot of Hams have already built and implemented the kit into their station setup; they use it every day for standard operations.

Software

The MDSR Team has released the MDSR V2.9 last fall. This release added notch filters and adaptive filters for SSB, CW and broadcast audio. The graphic user interface was also changed to allow for better navigation and to reduce the footprint on the screen. If the MDSR is used only for RX operation the window size can be reduced to only show the button and display for this task.

The post-processing filter options

Depending on the mode the MDSR is in, different filter options are displayed. This allows the optimization of each post-demodulation filter for better overall performance of the MDSR while receiving. In addition to a variable bandpass filter, one auto notch and one manual notch filter can be utilized to eliminate interfering signals.

Note: AM mode does not have an auto notch filter.

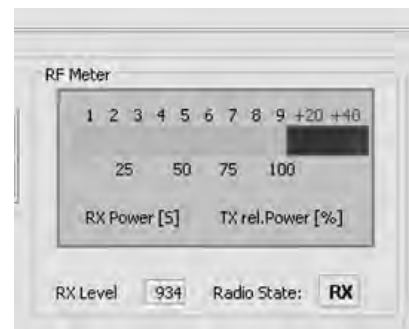


Version 3.0 was launched in February 2015. It received a totally new DSP engine to upgrade to the Jsyn 100% Java sound synthesizer. Now the MDSR core software can be adapted to all Java capable computers such as Linux, Ubuntu and Apple.

The MDSR V3.0 can be downloaded from our website and used for free for amateur radio purposes. For more info please visit our website or join the Yahoo user group.

New S- and Power Meter for the MDSR V3.0

The updated RF meter features a dual color display for RX. It turns blue while in TX mode. In receive the scale is in S units, and the meter displays Pout [%] during transmit.



The new “MDSR SA” Spectrum Analyzer

Introduction

Implementing the latest Jsyn DSP software in the MDSR V3 also gave us access to advanced signal processing devices such as FFT and IFFT conversion units. Our team started to develop a spectrum analyzer by using a spectral filter. This filter creates a 420 line audio spectrum (0 – 20 kHz) with a resolution of about 47Hz per line. After the first attempt to connect the SA to the IF of the LIF converter the result was encouraging, but the bandwidth response of the receiver’s IF looked like a “hump”. This made the detection of signals very cumbersome and useless for level comparisons.

The idea was born to compensate each spectral line individually so that the result was a flat spectrum. This is not practical, though, because it would take the user hours to adjust all 420 line levels individually. Sweeping the bandpass was another option, but this would require a RF sweep generator. Most amateur radio operators do not have access to such an instrument. Another option was to measure the receiver’s filter responses with RF noise, by tuning the radio to an unused section of the band and then taking a noise peak measurement for each frequency line.

The noise that is received on the antenna port is constant over a wide part of the spectrum. Therefore all the level differences are caused by the filter pass of the input and IF filters. So what looks like a “hump” – when noise is received – is really a flat spectrum. Now the DSP could be programmed to measure the peak level of each spectral line, calculate its attenuation and store the result in a table.

To obtain a flat spectral readout now only requires a simple multiplication of the corresponding attenuation factor by the current spectral input level. The result is a displayed spectrum that is flat. In essence the effects of the receive filtering, which are important for the radio's performance, have been mathematically removed.

How do Spectral Filters work?

Introduction

Spectral filters are a very interesting way to manipulate and create signals. The standard oscilloscope only measures voltages as the sum of electron pressure in conducting materials over time. It does not provide a frequency-separated display that shows the dynamic behavior of all the different groups of electrons that make up the electric current. These groups of electrons can move back and forth at different speeds (frequencies). The more electrons move in one direction, the higher the amplitude. Frequencies or tones are in essence oscillating electrons.

If tones are to be manipulated, it is much easier if the signal is separated by frequency (tones) than by time. Unfortunately it is much harder to display electrical behavior of a conductor in the frequency domain than in the time domain of an oscilloscope. This is also true in DSP (digital signal processing).

What is FFT (Fast Fourier transform)?

In DSP the frequency domain is so important that engineers developed special programs called FFTs that take streaming audio or video data and separate it into multi-channel data flows. FFTs can have 1000 or more channels separated by frequency; these are termed *bins*. These individual data flows can now be observed by frequency and amplitude for a specific behavior. A host program can display the data as a spectrum on a screen or make decisions if certain signal patterns are present. The applications of this technology in the radio field alone are almost limitless.

What is IFFT (Inverse Fast Fourier transform)?

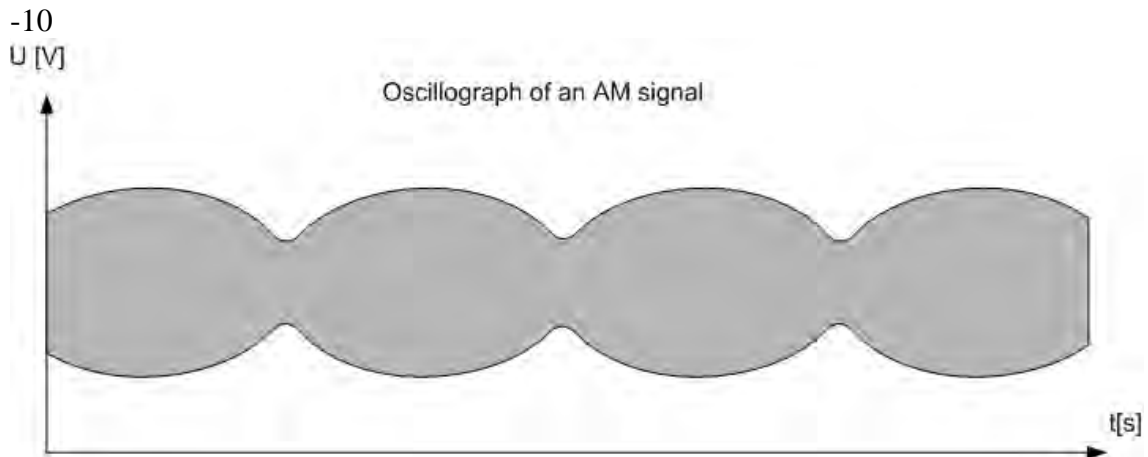
In DSP, IFFTs has an almost equally important role. IFFT and FFT are mostly used in pairs. Whereas FFT separates the input signal into different frequency channels the IFFT does exactly the opposite. It takes all the FFT bins – after they were processed – and combines them together into a time domain output that can drive a speaker after it has been turned back into an analog signal by a DAC (Digital to Analog Converter).

Applications

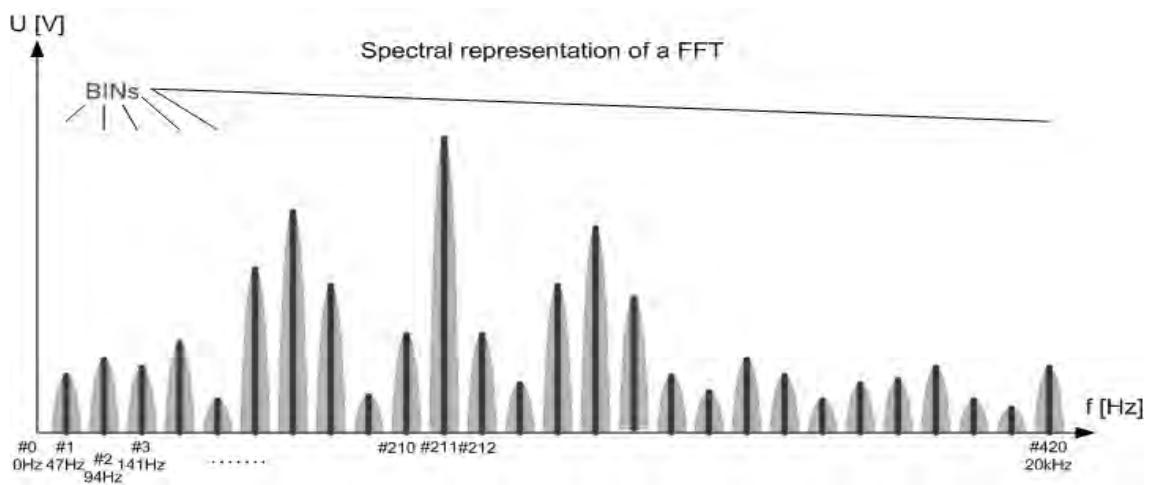
- Removing unwanted frequency bands
- Shifting frequencies or bands of frequencies up and down
- Removing unwanted carriers or static crackle
- Modulation and demodulation (digital or analog)
- Adding control tones
- Creating noise

Spectral Filter example with FFT / IFFT pair of 420 channels (BINs)

An AM signal is fed to the ADC and changed into a stream of numbers relating to amplitude.



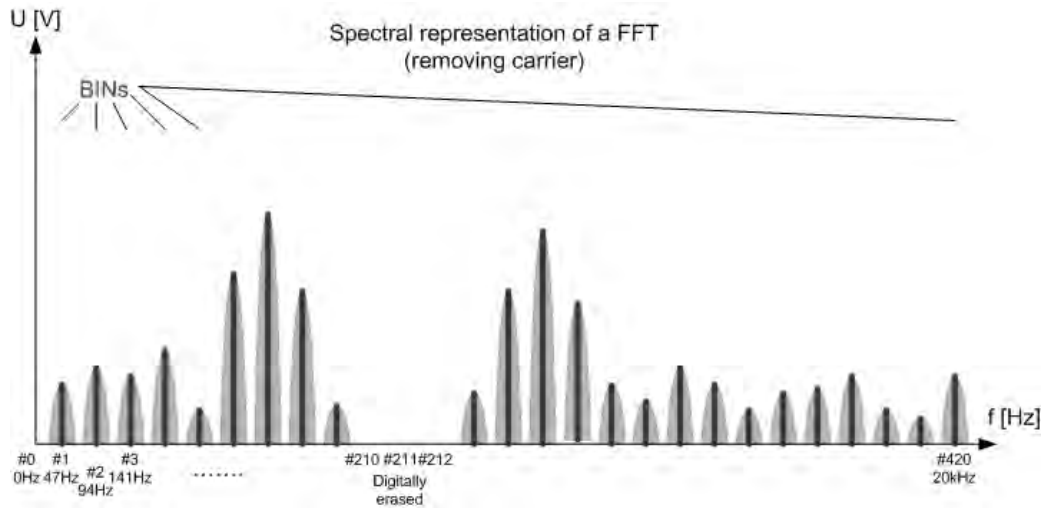
The now digitized signal is put through a DSP process called FFT



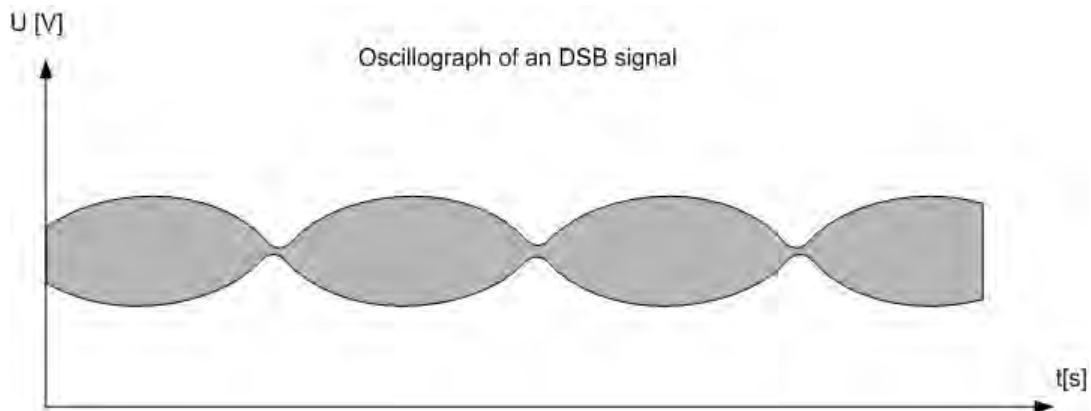
The FFT filters frequencies and stores the signal stream in BINs (table representation)

Bin #	0	1	2	3	4	5	6	...												420
time	a0	a1	a2	a3	a4	a5	a6	...												a420
000154	a0	a1	a2	a3	a4	a5	a6	...												a420
000155	a0	a1	a2	a3	a4	a5	a6	...												a420
000156	a0	a1	a2	a3	a4	a5	a6	...												a420
000157	a0	a1	a2	a3	a4	a5	a6	...												a420

Signal manipulation based on frequency (removing Bin 210 - 212)



Performing an IFFT (Inverse Fast Fourier Transform) and sending the data stream to a DAC will recreate the analog signal without the carrier.



Disadvantage of FFT filters

- All the data manipulation is processor intensive and delays are an issue. The program needs to reduce the amount of DSP data to the bare minimum before executing DSP functions. The software instructions need to be as basic as possible. Avoid coding that requires access to higher functions or peripherals such as keyboard, mouse or displays.
- The DSP processes can create huge amount of data. A 24-bit DSP chip with an FFT having 1000 channels and a sample rate of 192kHz creates 4.608MB of data every second!
- A fast computer is required to get the speed performance needed. Good coding can assist in reducing processor requirements.
- Signals can sound tinny and robotic due to the FFT- IFFT process. A good understanding of how to configure the DSP properly is a must.
- The high price of components and the availability of high resolution digital converters are limiting factors.

MDSR SA

Introduction

The new spectrum analyzer is written 100% in Java – including the DSP – and enhances the MDSR software on a Linux platform. Since it is written in Java it also will work with the existing MDSR installation in Windows.

One of the issues with viewing an IF spectrum from an analog transceiver via a sound card interface is the inconsistency of the IF amplitude and the narrow bandwidth. This makes the spectrum look like a hump with no signals on either side. To overcome this we have developed anew feature called SAC (spectral amplitude correction) which measures the attenuation of the IF and applies a correction value that greatly flattens the spectrum.

This document will also explain how to install and run the software in the above mentioned operating systems. A detailed manual is also provided.

This is a working document which will grow in size as we explore the possibilities of this new software. If you would like to join us and work on the continuing project, please register at the

MDSR Yahoo user group here:

<https://groups.yahoo.com/neo/groups/mdsradio/info>

Information on how to connect a receiver or transmitter to the soundcard can also be found at our website hosted by Guy Roels (ON6MU):

<http://users.skynet.be/myspace/mdsr>

Acknowledgement:

Special thanks go to Phil Burk for providing the Jsyn FFT sound interface free of charge for this project.

Installing on Windows with MDSR already installed.

Download the “Spectrum Analyzer. jar” from the Yahoo use group. Look in the file section in the “New SA for MDSR” folder. Copy the file into “MDSR” directory. Create a desktop shortcut. The program is started by double clicking the file or shortcut.

Note: If the spectrum analyzer is used without the MDSR, additional library files are required. Create a “lib” folder inside the folder holding the “Spectrum Analyzer. jar”. This “lib” folder and its contents can also be downloaded from the group.

Installing on Ubuntu (Linux), Apple Computer

Download the “Spectrum Analyzer. jar” from the Yahoo user group. Look in the file section in the “New SA for MDSR” folder. Copy the file into MDSR directory. Create a desktop shortcut. The program is started by double clicking the file or shortcut.

- Make the file executable by going into properties – permissions and checking the “Allow executing file as program”

Note: If the spectrum analyzer is used without the MDSR, additional library files are required.

Create a “lib” folder inside the folder holding the “Spectrum Analyzer. jar”. This “lib” folder can also be downloaded from the group. It contains three files that are required to make the program work.

Press the “Start” button to turn the spectrum analyzer on

Setting up the Soundcard or Device

The sound setup of the “New MDSR Spectrum Analyzer” is identical to the setup of the MDSR software. It always uses the default audio device, and can share it with other software at the same time. For more information on how to install and configure the LIF converter, go to the MDSR website or user group. If you have already set up the MDSR software, all the audio settings are identical. Both programs can work simultaneously without mutual interference.

Functionality of the Spectrum Analyzer

Introduction

This spectrum analyzer provides a span of spectrum for the audio spectrum of a sound card. It is designed to work with the MDSR software, but can also be useful for analyzing audio spectrum. It has a peak hold feature and a slow decay indicator.

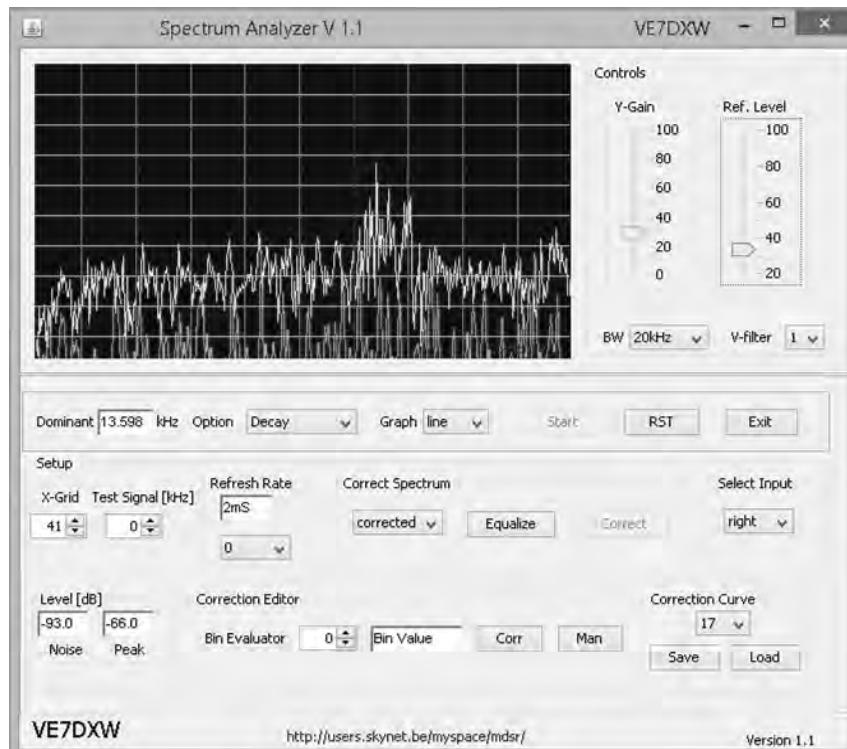
The new SAC (spectral amplitude correction) feature and how to use it will also be covered in this manual.

Updated in the new version V1.1

- **New line decay mode:** double spectral lines in “decay-line mode”.
- **Input Audio selector:** allows the user to select the left, right or both input audio channel
- **Storable correction curve files:** correction curves can be stored according to band
- **Spectrum analyzer stores previous settings:** for easy startup without calibration
- **Updated graphic user interface**

The new MDSR Spectrum Analyzer V1.1

Below is the graphic user interface for the new MDSR spectrum analyzer with the corrected display. The DSP removes the filter skirts which cause the spectrum to appear bent downwards at the edge of the filter passband. A flat spectral trace – like the one on a high-end lab spectrum analyzer – is displayed.



Correcting the filter skirts to obtain a flat spectral response

Correction Procedure

The correction procedure is a two step process where the filter noise response is captured and stored in the computer. Once the curve had been captured, it is applied to the spectrum and a normalized spectral display results. The filter noise response depends on the noise loading and the band the transceiver is on. The noise response can be saved and recalled for each band separately.

How to calibrate and correct the filter response in the MDSR SA

Depending on the speed of the computer used, this calibration will take about 10s or longer.

1. Tune the receiver on the specific band in use to an area that has no signals. Set the attenuators to 0dB and the preamps to OFF.
2. Under the "Corrected Spectrum" heading set the selector box to "normal". The spectrum changes back to the real display with the filter skirts (Fig. a).
3. Press the "Equalize" button and a display similar as seen below appears (Fig. b). The white dots indicate the flat response of the corrected spectrum. The green line represents the correction values. If the green line is not visible, move the "y-Gain" and/or the "Ref.Level" controls until it becomes visible. If there are signals present, they will be seen as large spikes on the filter curve. Tune to another spot without signals and repeat the measurement starting at 1. Small spikes, like the ones below, will be filtered out and a flat spectrum is displayed.
4. Press the "Correct" button to verify the correction line and to display the corrected flat display. Use the "y-Gain" and/or the "Ref.Level" control to adjust for best visibility.
5. If the spectral line is not flat, repeat the process.

Note: The measurement must be done with the antenna connected, so as to obtain band noise to measure the filter response.

Calibrate and correct the filter response in the MDSR SA using a white noise generator

The spectrum display correction curves can also be generated with the use of a wide band white noise generator, such as the MFJ-5014. To get the best results, the antenna noise level and the generated noise level have to be identical.

The output of the generator is connected to a step attenuator such as the MFJ-762 and then to the antenna input of the receiver. If the receiver has dual Antenna inputs, the RF noise can be connected to Ant 2, while the regular antenna is connected to Ant 1.

1. Connect the Antenna to the receiver and set the attenuators to 0dB and the preamp to OFF (Ant 1).
2. At the receiver, select the band to be measured.
3. In the "Level [dB]" box (bottom left) note the displayed level in the "noise" field.
4. Connect the receiver to the noise source (Ant 2).
5. Adjust the attenuator so that the displayed noise level is the same as that from the antenna.
6. Continue the above procedure at #2.

How to store the filter response in the MDSR SA for later recall

Once a flat filter response correction curve has been calculated and the corrected spectrum is flat, it can be stored on the computer's hard drive. Since the flatness of the display is dependent on the band and the noise loading, the MDSR SA has the capability to save one correction curve per band. Currently the MDSR SA stores curves for 160, 80, 60, 40, 20, 17, 15, 12 and 10 meters. After a flat spectrum has been calculated, it can be archived by following the above procedure for a specific band:

1. Under the "Correction Curve" heading, select the band that is currently in use.
2. Press the "Save" button. This will store a file containing the correction values.
3. Repeat the above procedure until all bands have a correction curve file.

How to recall the filter response in the MDSR SA

1. Under the “Correction Curve” heading select the band that is currently in use.
2. Press the “Load” button. This will recall a file containing the correction values.
3. The displayed spectrum is now corrected and should be flat.

Note: If the selected band does not have a correction curve, a warning message will appear.

Fig. a.) Spectral Display without correction (line mode)

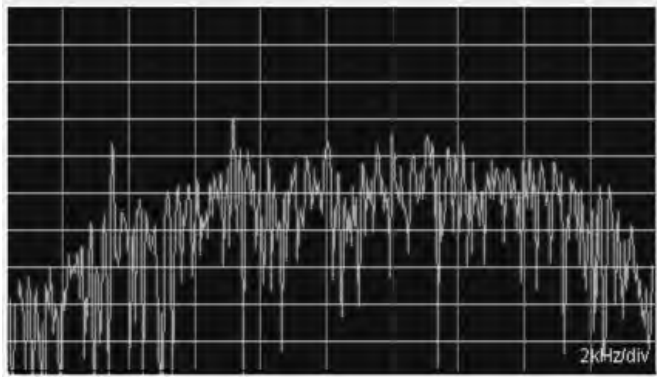


Fig. b.) Spectral Display while calibrating (line mode)

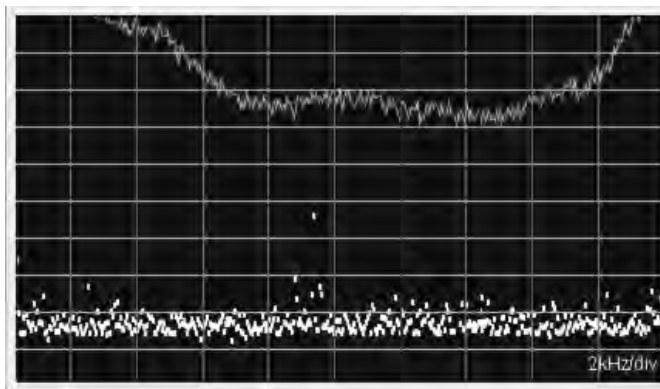
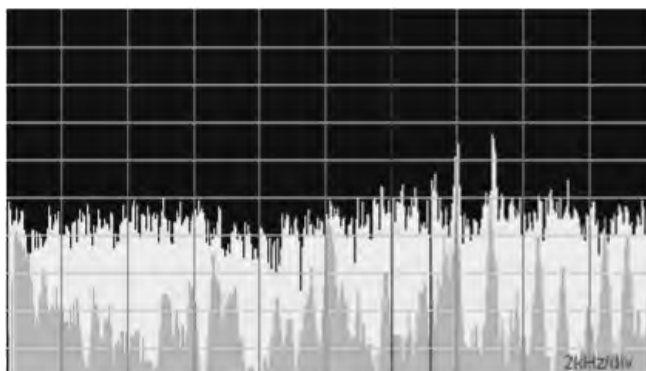


Fig. c.) Spectral Display after correction (bar graph – decay option)



Description of the user functions.

Introduction

This section describes the all the individual user controls and how they affect the displays and functionality of the MDSR SA software. Familiarization with their use and effects will allow you to adjust the display properly and to get the most out of this program.

Start Button

The “Start” button initializes the DSP and turns the spectrum analyzer on.

RST button

During operation, some sound devices build up a delay over time. This occurs, because the clocks of the FIFO data exchange buffers are not synchronized (a hardware issue). If there is noticeable delay, the “RST” button can be pressed to reset and clear audio buffers for lowest lag time.

Exit Button

The “Exit” button stops and removes the DSP engine from the computer and closes the program.

Dominant

The “Dominant” field displays the frequency value of the spectral line with the highest value.

When the spectrum is corrected, the highest amplitude may not be the dominant frequency.

Option (selector box)

The “Option” selection box has three options.

- Normal: (default) displays the spectrum as a single graph.
- Peak hold: displays the graph including its highest value. The highest value is held on the display indicating its frequency position.
- Decay: displays the normal graph including a time-lagged, delayed decay graph. This makes it easier to spot intermittent carriers.

Graph (selector box)

The “Graph” selection box allows choosing the line display or the bar graph display. The “line” setting is the default start up parameter.

Y-Gain

The “Y-Gain” slider changes the amplification of the displayed spectrum. It acts like a volume control and makes the spectral peaks longer or shorter depending on the setting.

Ref. Level

The “Ref. Level” control shifts the displayed spectrum up and down so that it can be placed properly in the grid display for easy readability. The control acts similarly to a fader on a stereo system. “Ref. Level” does not change the amplification of the display, but only its position.

BW (selector box)

The “BW” selection box provides three settings

- 20 kHz (default) provides a spectral display with 420 points at a resolution of 47 Hz. For the MDSR or IF display the 20 kHz selection has to be used to allow for proper placement of the red BFO line. The displayed spectrum is 0 – 20 kHz.
- The 10 kHz option can be used for audio processing and analysis. The spectrum displayed is 0 – 10 kHz. It will not center on the BFO line on the IF signal. It also has 420 display points with a spectral resolution of about 23 Hz.
- The 5 kHz option can be used for audio processing and analysis. The spectrum displayed is 0 – 5 kHz. It will not center on the BFO line on the IF signal. It also has 420 display points with a spectral resolution of about 12 Hz.

V-Filter (selector box)

The “V-Filter” (video filter) processes the spectral data and acts like a low pass filter to make the displayed spectrum smoother. There are five levels of filter intensity; 1 is the weakest and 5 the strongest. In the lowest setting (1) all the individual spectral lines are displayed. In the strongest setting the display lines are combined making it easier to see the spectral intensity of a SSB or broadcast signal. Default setting is 3.

X-Grid (spinner selector)

The “X-Grid” selector allows calibrating the spectral display with an internal or external signal source. The spectral display and the grid do not automatically line up, so it needs to be calibrated manually.

- Calibrating using the internal reference signal:
Set BW to 20 kHz, and turn on the “Test Signal” generator by setting the spinner control to 10. This puts out a white noise signal with sine wave peak at 10 kHz. Now, use using the “X-Grid” control, move the spectral peak behind the red center line. This will also calibrate the 5 kHz and the 10 kHz bandwidth selection.
- Calibrating using an external reference signal
The “X-Grid” control can also be used to be aligned with an external signal such as the carrier of an AM station, the radio’s calibration marker or a BFO. In this case the computer soundcard needs to be connected to a LIF converter and a receiver.

Test Signal [kHz]

The “Test Signal” selector allows adding an internal noise source and sine wave oscillator to the input of the spectrum analyzer. The test signal can be used to simulate the input for display or calibration purposes. Setting X-Grid (spinner selector) to 0 will turn the test signal off. Now the DSP is set to accept input from the audio default device.

Refresh Time

The “Refresh Time” default setting is “0”. This setting is the fastest, and if a P4 or higher processor is used the spectrum refresh rate is high enough to provide a nice, dynamically flowing spectral display. The FFT process needs a lot of processor time, and it can easily consume 60% CPU resources or more. This may affect other applications, and therefore a refresh time throttle is needed. Set the refresh time lower if other applications start to hang or are otherwise affected during the operation of the spectrum analyzer program. The refresh time indicator displays the time in mS for each sweep. Newer processors can be as fast as 1 to 2 mS.

Equalization (selector box)

The “Equalization” selector has three options.

- normal: the displayed spectrum is not corrected
- corrected: the display is normalized using the loaded correction curve.
- calibrate: this is used only during the calibration procedure – do not select manually.

Equalize button

The “Equalize” button should only be pressed after the receiver has been tuned to a section of the band without signals. During this operation, the DSP measures the filter response curve which can be viewed on the display.

Correct button

The “Correct” button activates the currently active (loaded) correction curve and corrects the spectral display.

Select Input

The input selector chooses the input channel of the default sound card. Three options are available (both, left, right).

Level [dB]

Under the “Level” label there are two fields that display the average noise level and the peak level per scan in dB relative to 1V_{pp}. These two variables are used to determine the bin with the highest amplitude, so as to calculate the dominant frequency value. The noise level indicator is also used for correction curve measurements.

Bin Value Evaluator

The "Bin Value Evaluator" consists of the bin value field and the bin number selector. The bin value field displays the correction value of the bin selected. This allows the user to view and edit stored correction values.

Note: A red marker bar is displayed in bar graph mode indicating the position of the displayed correction value.

Corr Button

The "Corr" (correct) button calculates the current displayed bin correction value by taking the adjacent values and averaging its position. This can fix miscalculated values which can cause holes or peaks in the spectrum.

Man Button

The "Man" (manual) button updates the manual entered correction value for the displayed bin.

Correction Curve

In the correction curve box, the option box selects the band that either saves or loads the correction curve data into the DSP. The "Save" button save the currently active correction data. To recall correction data, the "Load" button is pressed. The display is updated immediately.

Congratulations!

Now the new Spectrum analyzer for the MDSR is working and calibrated. Please follow the instructions and tune the new MDSR SA for best performance. We hope that you will have a great experience with the new spectrum display. If you have any questions or recommendations regarding this software or would like to get involved, please join us at the Yahoo user group:

<http://groups.yahoo.com/group/mdsr/>

Thank you for using the MDSR. Please tell your friends about this project.

All the best,

MDSR Team

June 2015