Installation, setup and brief operating guide for first time users of Linrad 04.08 with a Softrock receiver

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04.08 was the version current as of 20 December 2015.
I used Windows 7, but this guide should also work for Windows 8 and 10.

Linrad is an SDR program created by Leif Asbrink, SM5BSZ, that offers advanced features including options to change many FFT and audio processing parameters. This is great for those who understand such things, but it can make the setup appear daunting to those who do not (like myself). However, the setup process provides a "Newcomer" mode and default settings that can be accepted during the setup to obtain a working installation. This guide is basically an annotated log of what I did toinstall and set up the program and the basics of what I have learned about how to use it.

Almost all the information included here was obtained from Leif's webpages, videos, the program itself and several much appreciated email exchanges with Leif. Any errors are mine. I am learning about the program as I go along.

I hope that this guide will be helpful to others by saving them time setting it up and learning how it functions.

The audio quality that I have heard so far in my initial listening seems clearer and more natural than what I have heard from any other software that I have used with my Softrock. With a good transmitter/microphone/voice, the audio output sounds beautiful. In using this software I have the feeling that I am playing with a professional level instrument.

Note: It would be a very good idea to have the Softrock running with some other SDR software before trying to set up Linrad. This will ensure that both the Softrock and the sound cards have the appropriate Windows drivers installed and that they are working and controllable using the sampling rate and bit depth (e.g., 96000 kHz 16 bit) that you plan to use in Linrad. I had my Softrock Ensemble II working with HDSDR before I installed Linrad.

I am grateful to Leif not only for making Linrad available but also for his patience and willingness to help with questions that came up as I worked on this guide, and also to several others for their many helpful suggestions on earlier versions of this guide which have greatly improved the clarity.

Installation of the Linrad program

The following procedure may appear lengthy and involved, but in my case the installation and setup steps listed below actually went quite smoothly and everything worked properly without any issues. I estimate that about 30 minutes would be required for initial installation and setup in Newcomer mode, if just these instructions are used without reading material on the links. Additional time required for the I/Q balance will depend on equipment available.

There are installation instructions provided at

http://www.sm5bsz.com/linuxdsp/install/compile/wincompile.htm

however, the following should be all you need.

There are TWO required steps for the installation.

1. Install the DLL files.
As of 20 December 2015, there is a dll installer available at

http://www.sm5bsz.com/dll.htm

with the following link

setup-dll-02.exe (4273141 bytes)

and instructions:

"You may execute it without saving the download. It will create c:\linrad\dll and place the contents of linrad-dll-files02.zip there."

Execute this dll installer by clicking on the .exe file.

Note: if you choose to install the dll's manually, the c:\linrad\dll\ directory MUST exist (and the dll files must be in it), even if you run the linrad.exe main program file from some other directory. The program looks for the dll's in that location.

2. Install the main program. (A note on upgrading Linrad is at the end of the Guide.)

The Linrad home page

http://www.sm5bsz.com/linuxdsp/linrad.htm

has a lot of useful information about installation and setup of Linrad that could be reviewed before clicking on the link to the Windows installer that is included there and also copied below.

However, all you should need do is click on the following link.

http://www.sm5bsz.com/linuxdsp/archive/setup-linrad-04.08.exe

Execute this main program installer accepting all defaults.

Windows then gave an error message re. 'this program requires elevation'.

I ignored that and clicked on the new desktop icon to start the 32 bit version of Linrad (the 64 bit version also worked fine).

On another installation when I tried to start the program the first time the main

screen opened but it hung. I closed the window, right clicked on the desktop icon and clicked on "Run as administrator", then it ran OK. You might have to argue with Windows.

Linrad started, showing a

"WELCOME TO LINRAD This message is not an error ..." screen.

This screen begins the "Global Parameters Setup" procedure.

Initial Setup

n, enter   to invoke Newcomer mode

n, enter   to enter screen size in %

I entered 85 for both height and width.

85, enter

85, enter

A new menu screen then appears that should now have a line at the bottom:

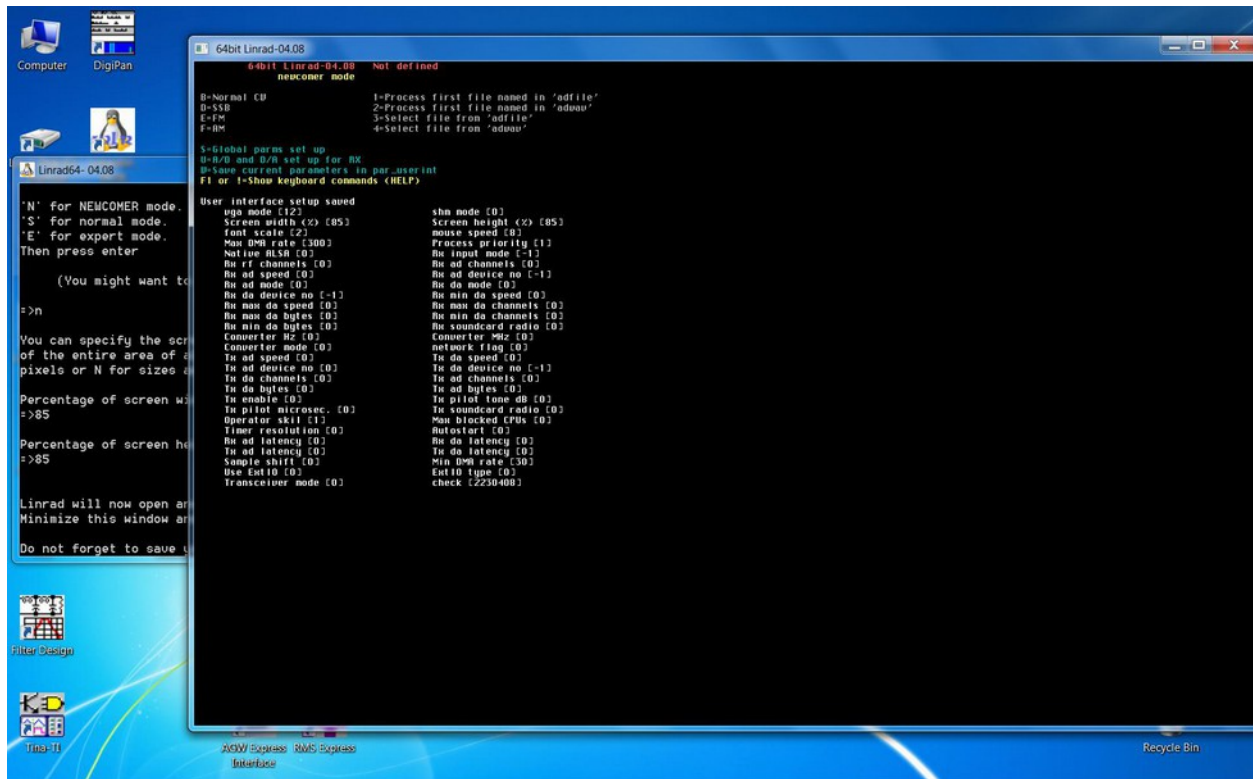"The required DLL package is installed"

(if not, the DLL folder is not in c:\linrad.)

This is the main Linrad menu.

w (no enter)

This should result in a long list of parameters, in white text, being displayed.

Here is how my screen looks at this point.

It "w" does not produce this, you may (as I did) need to click on the other window still open, shown to the left on the screen shot, then go back to the main menu screen and type "w" again. **Note: this other window was originally hidden under the main window, so you may need to look for it.**

Using instructions in the "Linrad for Newcomers" page

http://www.sm5bsz.com/linuxdsp/usage/newco/newcomer.htm

u (no enter)   to enter A/D setup

a (no enter)   to enter hardware selection

a (no enter)   to select soundcard input option

It now asks whether or not to use Portaudio, which is soundcard control software.

(For some information on this, Leif gives results for testing of one soundcard for receiver input under various versions of Windows, with and without Portaudio.

http://www.sm5bsz.com/linuxdsp/hware/sb-live-usb/sb-live-external.htm)

To avoid any unneeded complications, I decided not to use Portaudio for an initial setup of soundcard input.

n (no enter)   to not use Portaudio

It then gave a list of sound cards, in my case the first one was the one I wanted to use.

0, enter   (You should select the appropriate choice for your system.)

It then asked for sampling speed. I wanted 96000, so I entered

96000, enter

Note: it did not ask for number of bits. The later summary screen reported it was using 16 bits. "Normal mode" (see below) did ask for selection of 16 vs. 24 bits. Another user has told me that in his setup Newcomer mode did ask for number of bits, so this may depend on your soundcard or driver, and of course on whether the card supports more than 16 bits.

It now asks for "select radio interface". I selected "2", "One RF, two audio channels (direct conversion)", which the F1 help key said was for I/Q input.

2 (no enter)

It now asked for "Number of points to time shift between I and Q". I selected -1.

(The value of -1 was determined by trial and error. I first put in +1 as on HDSDR for this soundcard I had to use +1 on the right channel to get the image null to work. With 0 or +1 there was very little image rejection before calibration, with -1 there was some. (Another of my cards requires "0".)

-1, enter

It now asked for "Select receiver hardware". I selected "3, Si570".

3, enter

It now presents a summary of current A/D D/A settings and asks for RX output to be selected with menu option b.

b (no enter)   to set up soundcard output

It now asks whether to use Portaudio for output.

   http://www.sm5bsz.com/linuxdsp/linrad.htm

gives a list of soundcard options in order of performance for the output side.

1: Portaudio ASIO
2: Portaudio WDM-KS
3: Portaudio WASAPI
4: Native MME5
5: Portaudio Direct Sound
Do not use Portaudio MMe.

and recommends to:

"Select the best (first) alternative from the list that exists for the soundcard that you want to use for output. The difference between these drivers may be very large."

While this indicates that Portaudio provides better performance for output, I decided to stay with native Windows MME for the initial setup of Linrad and plan to experiment later with the Portaudio options.

Thus, I entered

n (no enter)   to not use Portaudio

It then gave a list of sound cards. In my case the first one was the one to use.

0, enter   (You should select the one that is appropriate for your system.)

It then presents the A/D D/A setup settings summary and further options.

I selected "d" for "set parameters for soundcard radio hardware".

d, no return

It then presents a summary screen "Current setup of si570 usb device". I knew of no reason to try to change anything.

x (no enter)   to exit

It then presents a summary screen for "Current A/D and D/A setup for RX"

x (no enter)   to exit

It presents a reminder to save by pressing the W key on the main menu, and says "press any key"

I pressed the down arrow key

It presents the main menu.

I pressed

w (no enter)   to save the setup profile. **This is an important step.**

It says "User interface setup saved" and lists a summary thereof.

I then tried to start Linrad running in the CW operating mode.

b (no enter)   to select "normal cw".

It said "par_cw file missing, press any key".

I pressed the down arrow key.

It now presents a menu to select options.

This screen is shown at

http://www.sm5bsz.com/linuxdsp/usage/newco/newcomer.htm#INSTALL

under "Mode parameters. First time"

I used the mouse to move to "Continue" to accept the defaults and clicked (alternatively just press enter).

It then presented another menu, this is also shown in

http://www.sm5bsz.com/linuxdsp/usage/newco/newcomer.htm#INSTALL

under "The last mode parameter screen", which suggests to accept the defaults.

I moved the cursor to "Continue", and clicked (alternatively, just press enter).

The Linrad operating screen appeared and the radio seemed to be running!

(On one of my instals, a Windows error message appeared at this point saying the program might require administrator privileges, I just restarted the program and it ran OK.)

There was a waterfall, under that a spectrum window, and several other windows whose function was unclear to me at first.

Click on the spectrum or waterfall to select a frequency and start the radio. The waterfall and spectrum displays will start moving and sound should come from the speakers.

Note: if nothing is heard on the speakers, make sure that the "RF gain" control is not muted. There is a window with "Reserved for noise blanker" at the bottom. In this window on the far left is a vertical scale. Clicking on the top of this scale should turn the scale red. Sound should now be heard. If not, check that the mixer gain for the soundcard output device is set high enough.

Before trying to do anything else, consider the following.

Help function

http://www.sm5bsz.com/linuxdsp/usage/newco/newcomer.htm#INSTALL
gives the very useful info:

*"Place the mouse on an empty space outside the windows and press **F1**. This will show where all mouse controls are… All control areas are rectangular. Three sides become green and the fourth side red when F1 is pressed with the mouse placed where no mouse function is defined. The red side moves slowly counter-clockwise around each control area. Control areas are generally within a window, but an entire window or parts thereof may also be a mouse control. The border lines are also mouse controls.*

*Press any key to return to the normal processing screen. Then place the mouse inside one of the control areas (or a border line) and press **F1** again."*

It is best NOT to click on anything or change anything before finding out what it does. The operating screen provides almost no labels for anything. This gives a nice uncluttered view once you have learned what things do, but this is perhaps not a place to try to learn by just pushing buttons. **Use F1 before clicking if not sure what will happen.** Writing down what the default was before you change it would also be a good idea.

This link provides a summary of the screen controls, generally listing the information provided by the F1 help key.

   http://physics.princeton.edu/pulsar/K1JT/Linrad_On-Screen_Controls.pdf

Confirming that it is working

There is a small "Center Freq" window. Clicking on that allowed entering a new freq. This is the Local Oscillator freq. as when I entered 14.0, I could hear the Softrock LO on 14.000 on a nearby receiver.

I connected an external antenna to the Softrock and the noise level increased. A small vertical rectangle on the right with "S" numbers showed higher levels with the antenna connected.

At this point you should check to see whether you need to reverse I and Q. Tune a signal on a known frequency. If you have a signal generator you can use that. Since I do not, I used WWV.

I pressed x several times to get back to the screen that allows changing mode to AM for WWV reception.

I got to that and pressed "f" for am.

This produced another "parameter selection" screen, "par-am file missing, press any key".

I pressed the down arrow key.

Next screen, I selected "continue" and clicked.

Next screen, selected "continue" and clicked.

The main operating screen appeared in AM mode.

I changed to Center Freq of 10.010 and could see WWV on 10.0, but the image on 10.02 was stronger. This means that I needed to reverse I/Q. Clicking on either signal gave clear audio. (With all of my Softrocks, HDSDR and SDR-Console both require selection of "Swap I/Q", so I was not surprised when it was needed also in Linrad.

Backup the installation (this is optional).

At this point I decided to do a backup before changing anything else as I had a working system.

I pressed "x" a couple times to get back to the main menu, then

w (no enter) to save parameters

Then I copied the entire c:\linrad directory and saved it in another location.

In case I messed something up with subsequent changes I wanted to have a copy of this working system that I could restore to the main directory if needed to avoid going through the setup procedure again.

Swapping I/Q

If you need to swap I/Q, you have two alternatives.

1) Change the soundcard cabling. If your soundcard line in has separate jacks for left and right, simply reverse the plugs. If the soundcard has a stereo jack, you can rewire the plug to swap the hot wires for left and right. This is what I did.

2) Swap I and Q in software. The way to do this in Linrad is, from the main operating screen, press:

x (no enter)	shows parameters
x (no enter)	main menu
u (no enter)	A/D and D/A set up for RX
d (no enter)	Set parameters for soundcard radio hardware
h (no enter)	Change PASSBAND DIRECTION
-1, enter	to "reverse the spectrum displayed on the waterfall graph"
x (no enter)	
x (no enter)	shows a reminder to save with W, and "press any key"
I pressed down arrow	
w (no enter)	to save parameters
f (no enter)	for AM mode (or d for ssb, etc.)

Google user group for Linrad

I was unsure how to proceed with the I/Q swap and image nulling so I went to the Google user group to ask for help.

> https://groups.google.com/forum/?fromgroups#!forum/linrad

W3SZ replied with a load of good info including this link

> http://www.nitehawk.com/linrad_dat/linrad_dl_man.pdf

which is a 94 page instruction manual for Linrad. I believe this was prepared by DK1VI (although the title page says "LINRAD by Leif Asbrink, SM5BSZ") but which is largely a compilation of sections from several of Leif's webpages. Although slightly out of date, it is still very useful.

W3SZ pointed me to the info for image nulling:

> http://www.sm5bsz.com/linuxdsp/iqbal/iqcal.htm
> http://www.sm5bsz.com/linuxdsp/iqbal/iqbal.htm

See also page 27-32 of the above pdf manual.

I/Q Image balancing

Note: do not attempt to run the I/Q balancing routine until you are certain you have the "Number of points to time shift between I and Q" parameter set correctly. If it not correct, the routine will not be able to null the image. If it is correct, you should see the image signal strength lower than the main signal strength even before you have calibrated. If it is reversed, or equal, something is wrong. (If you think you had the shift correct, and the routine does not reduce the image significantly, re-consider the shift parameter.)

Linrad can compute a correction function for many segments across the displayed spectrum (e.g. 96 kHz for 96 kHz soundcard sampling rate), not just at one specific frequency. This provides for excellent image rejection across the whole observed spectrum.

Doing this requires providing strong stable signals at many frequencies distributed along the observed spectrum. The links below show this being done with a lab quality signal generator connected to the receiver.

Leif has a video that includes a section describing I/Q balancing.

https://www.youtube.com/watch?v=PLISi8iq_LE

An illustrated text describes the procedure

http://www.sm5bsz.com/linuxdsp/iqbal/iqcal.htm

The procedure is quite neat. The program displays the spectrum range, showing the noise floor. You then provide a strong RF signal, which shows on the display as a spike in red, which then changes to green as the program calculates the correction for that frequency to cancel the image. You then step to another frequency and repeat until the whole spectrum is green, then you save the correction function.

As I did not have an RF signal generator, I used a conventional HF transceiver transmitting a low power signal to an antenna, with a short wire connected to the Softrock as a receiving antenna.

I set the Linrad center frequency to 14.100 (I used SSB mode) and began with HF transceiver set for CW on 14.105. On the normal screen showing 14.060 to 14.140 or so, when I briefly keyed the transceiver I could see the signal on 14.105. Then press:

x (no enter)   to bring up options

c (no enter)   to select the calibrate option

a (no enter)   to select I/Q balance calibration

c (no enter)   to clear RAM (this initializes the segments I believe).

It now shows a spectral display and is ready to compute corrections at multiple segments. (The default is 32 segments, which is plenty, but the menu also gives you the option to change the number of segments. If the sampling rate is an exact multiple of the number of segments, such as 32 segments for 96 kHz sampling rate, this makes for a convenient integer number of kHz per segment step size for adjusting your generator frequency.) This display is basically a horizontal line of white dots with a peak in the center (the center hum spike that is visible on the main spectrum).

I transmitted a 5 W signal on an external antenna and listened to the Softrock with an internal antenna wire of length to give a strength on the A/D line of about 50%, which one of the instruction references said to use.

I then keyed the transceiver while watching Linrad. The signal appeared as a peak on the line of dots with the dots forming the peak initially all red. These turned to green in less than a second and the display reported "Segment X OK". (If your PC is much faster than mine, you might not see the red color. The important thing is that the white dots change to green.) I moved the transceiver up a few kHz and keyed again. This time a new peak appeared, just to the right of where the green segment was, again first in red then turning to green. I repeated this, moving up in frequency in steps until the whole right half of the line of dots was green. **Note. if you reversed I/Q in software, the peaks move to the LEFT when you shift frequency UP, this is OK**. After getting the one half green, repeat the procedure using signals below the center frequency for the other half and finally go back and hit any segments that are still white until the whole line of dots is green. The segment numbers may not change in regular order.

Once this is done, press:

u   to compute new calibration and store in RAM

Now press enter, this is important. (I think ENTER is necessary to actually store the calibration, but even if not, you want to view the curves to confirm that the calibration has been computed.)

It now shows the calibration function and gives instructions on how to zoom the display. There is no need to zoom.

s (no enter)   to save current RAM contents to disk.

This is a necessary step. If this is not done the calibration is not stored and the image rejection is not changed and you will have to do it all over again.

You now can exit back to the main menu (press x twice).

After I did this, images were invisible, with greater than 70 dB cancellation.

You may need to do this separately for different bands. (Once I get my configuration all set up I intend to make multiple copies of Linrad, one for each band where I do the IQ calibration, so each copy will have images nulled optimally for its band.)

According to

http://www.nitehawk.com/linrad_dat/linrad_dl_man.pdf

page 29, the I/Q calibration procedure can be run multiple times to improve the rejection.

"In case the system is already calibrated and you only want to refine to absorb effects of ageing or temperature differences, press plus and then minus to clear collected data while keeping the old calibration."


I did this I/Q calibration separately for each mode, CW, SSB, AM.

However, Leif has explained that you only need to do this once for one mode and then can create copies of the calibration file and rename them for the other modes. For example, if you had done the I/Q calibration in AM mode, then *"You can copy DSP_AM_IQCORR to DSP_SSB_IQCORR and DSP_CW_IQCORR"* so that the calibration will also be used for SSB and CW modes.

Linrad is now installed and setup in "Newcomer" mode. It should be useable.

The various windows can be resized and moved to customize the display by dragging the borders of the windows. I did that to produce this screen layout, which shows the

radio running in SSB mode on 20 meters. (This is for version 04.07 but 04.08 should allow the same display.)
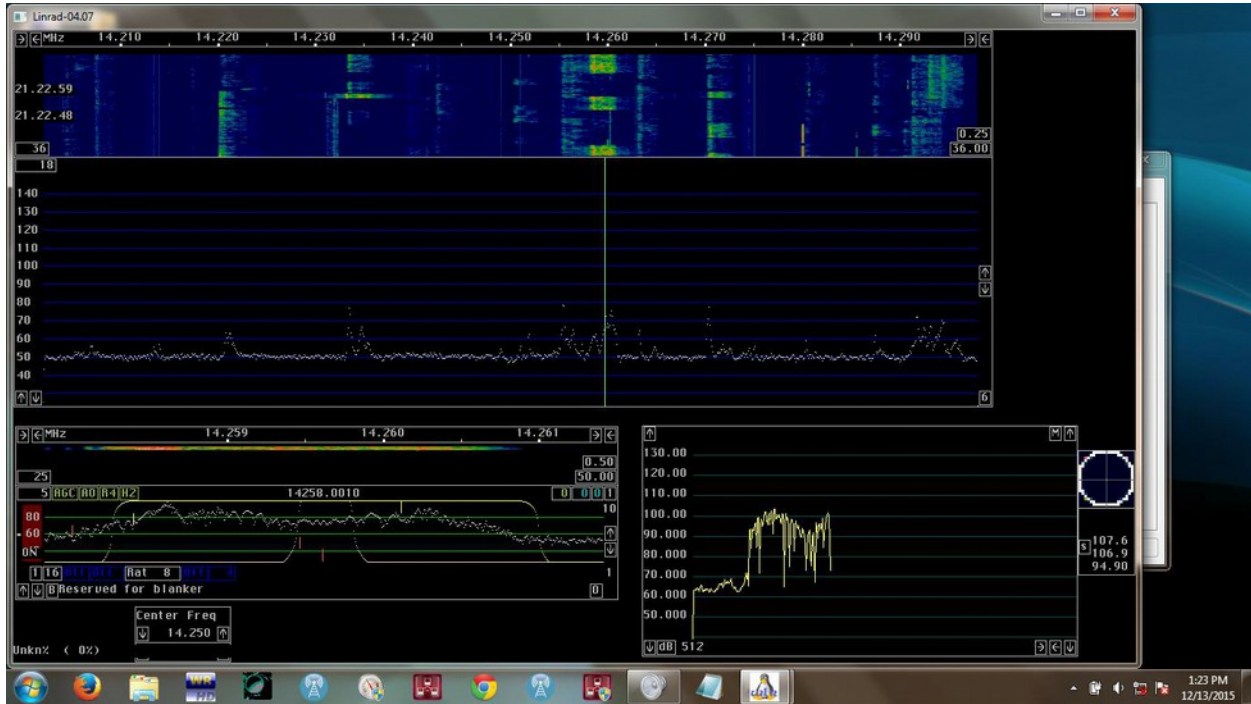
Operating the radio

Now that the radio is running, in addition to changing the mode (CW, SSB, AM, etc.), which is done from the main menu, we need to be able to control the frequency, volume, filter bandwidth, USB/LSB, and be able to read signal level. You might just want to play with it now and discover how to run it yourself, which is quite fun to do (**using the F1 key first before clicking on things),** otherwise, the following might save time.

Frequency control

The local oscillator frequency is set by typing a number in the small "Center Freq" window.

Unlike other software like HDSDR, in Linrad one can not select an exact listening frequency by keyboard entry or by clicking on digits to increase/decrease the "tune frequency". Like other software, the listening frequency can be roughly selected by clicking on the main waterfall or spectrum. In Linrad this can then be shifted in steps by clicking on left and right **keyboard** arrows or by using a mouse wheel. The step size is controlled by the + and - keyboard keys. You can reduce the step size to the single Hz range if desired. You can also move the frequency a small amount by clicking in the baseband waterfall or in the zoomed ("high resolution") spectrum window. You can also drag the frequency bars in either waterfall with the mouse to tune.

I have found that the Linrad tuning controls can be understood by keeping in mind that there are three properties being controlled, each one can be shifted relative to the other two.

a) the frequency that the radio is "tuned" to. (For a properly tuned AM or SSB signal this would be the frequency of the carrier or suppressed carrier.

This "tune" frequency is displayed (in .1 Hz resolution with the default FFT settings) in the middle of the baseband window, directly above the displayed filter passband.

b) the location of the filter passband in the spectrum. When one clicks on a signal on the waterfall or spectrum, the passband is moved to straddle that location. This is unlike other SDR programs for which clicking on the waterfall moves the "tune" or "carrier" frequency to that location, with the passband extending to one side or the other. (For SSB the tune frequency is normally not in the passband, for CW and AM the signal carrier would normally be in the center of the passband.)

A signal within that passband is shown as a peak in the spectrum display and is also visible in the waterfall above the frequency readout display line. Since the current passband width and location is also shown, you can easily see where a carrier or CW signal is relative to the passband.

c) the offset of the BFO (Beat Frequency Oscillator) relative to the tune frequency.

This offset produces the audible pitch tone of the CW, e.g. 500 Hz.

The left/right keyboard arrows can be used to do any one of three things according to which tune mode (**T, P, B**) is currently selected.

The current tuning mode is shown in the small box to the left of "Reserved for noise

blanker" text in the baseband window. These three tuning modes can be selected among by pressing the M keyboard key (or by clicking on the box that shows the current mode).

This is what seemed to me to happen in each tuning mode.

1) **T** Shift both the **T**une frequency (the frequency readout changes) and the passband, locked together, while the BFO frequency maintains a constant offset from the tune frequency. This would be the normal mode for tuning around the band and is like what happens in a conventional receiver when one turns the **T**uning dial. The pitch of a CW or SSB signal changes as you tune. (The F1 help key for the tuning box says: "T = Tune center frequency".)

2) **P** Shift both the tune frequency (the frequency readout changes) and the **P**assband, locked together, while the BFO frequency is shifted in the opposite direction to maintain a constant offset to the signal. This is like what happens in conventional receivers when the **P**assband control is rotated. Pitch does not change, but the tuning range is limited since once the passband is moved beyond the desired signal it can no longer be heard. This could be useful in CW to shift the passband to drop an interfering signal off the bottom or top of the filter. (The F1 help key for the tuning box says: "P = Tune passband".)

3) **B** Shift the **B**FO frequency while leaving the tune frequency and passband fixed. This is like the **B**FO control in older receivers. It changes the pitch of a CW signal while maintaining the signal at the same position within the passband. (The F1 help key for the tuning box says: "B = change BFO frequency.)

For SSB, you can first tune the frequency roughly by clicking on a signal or frequency in the waterfall. In Linrad you click on the centre of the waterfall signal, not on the lower (for USB) or upper (for LSB) end of the signal as in other programs. Then (press 'M' if required to show the T tuning mode) fine tune using the L/R keyboard arrows or mouse wheel for correct pitch.

For USB vs. LSB, see the next section.

Filter width and shape, BFO location vs. passband, USB/LSB

In the baseband window that shows the filter, there are two small vertical yellow bars. Dragging the top bar changes the width and dragging the bottom bar changes the skirt sharpness of the filter. The three small vertical red bars can be dragged to move the BFO frequency relative to the passband. The top red bar moves the frequency slowly. The two lower ones move it progressively faster. **Moving the top red bar all the way to the other side of the passband is the method used to switch between USB and LSB.** There is no direct readout of, or method to directly specify, the filter width, nor the offset from the BFO to the passband.

In SSB mode, the filter width and offset values can be obtained by using the BFO frequency readout above the filter. In SSB mode this is the tune frequency. One can move the BFO frequency to the upper and lower edges of the filter passband, noting the frequency of each, and subtracting to get the width. The offset can be set by moving the BFO from the edge of the passband, noting the frequency, and then moving the BFO frequency the desired distance from the passband.

In other modes the indicated frequency is the passband center.

While the above is unlike other software I have used, in practice one just listens to the audio while adjusting the filter width and offset to obtain the optimum intelligibility.

Volume and AGC controls

There is a window that shows the filter and a zoomed in spectrum. Leif calls this the "baseband graph". There is a small vertical red bar on the left side of this window. This is an "RF gain control". I set it to maximum by clicking on the top of the scale. There is an AGC button (on/off) and controls for setting the attack, hang, delay properties.

Linrad has no separate audio volume control. You use the Windows mixer volume control for the device that was selected for audio output to control the sound volume to speakers or headphones.

S meter

W3SZ explained:

Click on the little "S" in the lower left corner of the coherence graph and it will bring up the S meter. Clicking on the dBm label in the lower left hand corner of the S meter will toggle between dBm and S units. Clicking in "P" or "M" or "2" in the upper right of the S meter will toggle between peak and RMS and both. The coherence graph is the little rectangle with vertical long axis that has a cross-hairs with a dot/blob/circles in the center on top and 3 lines with numbers below. The little S that you have to click is to the left of the numbers on the second line. In case you don't know what the coherence graph looks like, the coherence graph [and the S meter] are at the right edge of the screen here:

http://www.nitehawk.com/w3sz/showcoherencegraph.gif

The S meter is initially a thin vertical rectangle which shows signal strength as a bar. If one drags the edges to make a horizontal rectangle, the display changes to a moving point where the signal strength is indicated by the scale on the left side. There is also a number showing the instantaneous value of the signal in the "coherence graph" window. The moving point version is very useful for test purposes, like comparing the strength of the main carrier to intermodulation spur levels.

Waterfall speed and contrast

To increase the waterfall speed, decrease the number in the box in the lower left corner of the main waterfall graph.

This number is "Waterfall avgnum" (default=36 for CW and SSB modes in my initial **Newcomer mode** setup). The default value of 36, together with the FFT1 defaults, provides a very slow waterfall. Changing Waterfall avgnum to 6 provides a much faster speed, more typical of what is seen in HDSDR. The minimum allowed value in this box in Newcomer mode is the value in the lower right box of the main spectrum window. If the latter is changed to 1, then 1 can be entered in the Waterfall avgnum box, and the waterfall becomes fast enough to read CW visually. Waterfall speed also depends on the FFT1 size. If the FFT1 resolution is increased (e.g. to 50 Hz instead of 100), the waterfall slows down.

In **Normal mode** with FFT2 enabled ,"1" could be entered for "Waterfall avgnum" regardless of the number in the lower right of the spectrum window, but the waterfall speed was slower than in Newcomer mode for the same settings. If a still faster waterfall speed is desired, FFT1 resolution can be decreased, e.g. to 200 Hz or greater, or "Second FFT bandwidth factor in powers of 2" can be reduced from the default 2 to 1 or 0. These are done by going to the parameters menus, from the operating screen press X, then P. In

http://www.nitehawk.com/linrad_dat/linrad_params_1.html

Leif recommends setting the latter to 0 or 1 for SSB. Changing from "2" to "1" appeared to double the speed of the waterfall.

To adjust the waterfall color for optimum visibility of weak and strong signals, change the numbers in the two boxes at the lower right of the main waterfall graph.

The top box, "Waterfall Gain" controls contrast, and the bottom box, "Waterfall Zero" sets the brightness at the noise floor (minimum signal level).

Noise blankers and setup in Normal mode

It is not necessary to continue below this point.

You only need to do so if you wish to enable the noise blanker function or need other features only available in "Normal" mode, including setting the noise floor to the recommended level.

Copy the Newcomer mode system to a new directory.

In order to use the noise blankers, you must first initialize "Normal" mode instead of "Newcomer" mode. This mode provides many more options and thus many more opportunities to mess it up.

At this point I decided to create a new instance of Linrad that I would change to Normal mode, while leaving the working "Newcomer" mode system in the original directory alone.

I pressed x a couple times to get back to the main menu, then

w (no enter) to save parameters and then exited from the program with ESC.

I then copied the entire c:\linrad directory and saved it in another location. I also created a desktop shortcut to linrad.exe in the new location.

I then restarted Linrad from the new location.

Change the new Linrad to Normal Mode.

From the main menu,

s (no enter)   to enter the Global parms set up menu

n (no enter)   to change to NORMAL mode

n (no enter)   to stay in NORMAL mode

It asks about font scale, I chose "2" ("1" is smaller, too small for my screen, 5 is very large)

Note: if you go back and change this, you will have to restart the program for it to take effect.

2 (no enter)

It asks about process priority, I chose normal

0 (no enter)

It now asks "How many [processors] do you allow Linrad to block?"

Leif explained that if "0" is selected, Linrad's FFT processing can be interrupted by other programs' requests for CPU time, whereas if you need lots of CPU for Linrad and can not tolerate random time delays, you can reserve some of your processors for only Linrad. I chose "0".

0, enter

It now asks about screen size.

Continue with the setup procedure as had been done at the start of Newcomer mode.

It was not necessary to select the soundcard options this time, but when I did, in the soundcard selection process there were two new questions presented after selecting the input soundcard.

"Do you need more channels from the same soundcard? (Y/N)". I said no.

n (no enter)

"Use extended format"? I said yes. (I believe this allows using 24 bit sampling if the soundcard supports this.)

y (no enter)

After asking for the sampling rate, it now asked "No of bits (16/24)". I chose 16.

16, enter

Note: I later selected 24 and the summary said it was running in 24 bit mode, but I saw no difference in noise level ("noise floor", see below) with my noisy sound card. So I went back and re-entered the soundcard setup and changed it back to 16 bits to lessen processing demand.

Make sure it is running like it had been in Newcomer mode before continuing on to change anything.

Enable the Noise Blanker

Note that the following needs to be done for each mode, e.g. SSB, Normal CW.

When in the operating window with the spectrum and waterfall,

x (no enter)   to go into the options menu

p (no enter)   to go into the parameters menu

Now, in the Normal mode, there are more options available to select.

Mouse to "Enable second FFT", click, and enter "1".

1, enter

This is all that is needed to have the "dumb" noise blanker work.

Then click on "CONTINUE" (or just press enter) four times, until you are back in the operating window

The screen will now show the "dumb" noise blanker box (it has a green bar at the bottom).

There are other options to optimize performance.

See the noise blanker sections in

http://www.sm5bsz.com/linuxdsp/usage/newco/newcomer.htm#INSTALL

and

http://www.sm5bsz.com/linuxdsp/blanker/fp/fpblank.htm

Setting input levels

Leif has a general guide for setting up device gains for optimum sensitivity and dynamic range at

http://www.sm5bsz.com/linuxdsp/hware/alevel.htm

I suggest you consult the above first.

What I did as a first attempt is described next.

The noise floor needs to be checked and gains set.

This needs to be done for each mode, e.g. normal CW, SSB.

The instructions for this are in

http://www.sm5bsz.com/linuxdsp/install/dlevel.htm

especially the section

http://www.sm5bsz.com/linuxdsp/install/dlevel.htm#PRACTICAL

Start Linrad in normal operating state (showing waterfall, spectrum) connected to the Softrock, e.g. in SSB mode, ideally with a dummy load connected (less ideally with just no antenna connected).

Type "A", this opens a small window in the lower left corner.

Look at the number after "Floor". This is the noise floor.

According to the instructions, the desired value for this is in the range of 10 to 30 dB with the system running with a dummy load connected to the receiver.

In other places, Leif has recommended that the noise level should rise about 10 dB when the Softrock op amps are powered (i.e., when the 12Vdc is connected). That is, the noise level of the soundcard analog to digital converter should be 10 dB lower than the Softrock's output noise with dummy load.

This is possible for my "pro audio" external USB soundcard.

For this one, with the 12V plug was pulled, the noise floor was about 3 dB and rose to 14-20 dB (depending on band) when the power plug was plugged in. The default first FFT amplitude of 2000 was left unchanged. (I had to reduce the soundcard line input gain a bit, otherwise the noise increased too much when the power was applied.)

However, my $20 96 kHz card that has a much higher noise level, higher than the Softrock's output noise level. For this one, the noise floor did not increase when the 12V

was applied regardless of the soundcard input gain setting. I had to reduce the first FFT amplitude to 500 to get the noise floor down under 30 dB.

I reduced the First FFT Amplitude as follows.

From the main operating screen,

X to exit to the first options menu, this shows the current parameters,

P to change parameters,

mouse to the "First FFT amplitude" line and click.

Enter a new value (I tried first 1000, went on to check the noise floor, then repeated the procedure setting to 500). The 500 value brought the noise floor down to 12 dB. A lower noise soundcard would likely not require using the 500 value.

Click on Continue four times (or press return four times).

This brings you back to the main screen.

Type A to see what the noise floor value is after changing the FFT amplitude.

N.b., there is a second "smart" noise blanker. Setting up for this requires the use of a laboratory RF pulse generator.

The procedure is shown in:

>   https://www.youtube.com/watch?v=PLISi8iq_LE

As I do not have a pulse generator, I was not able to do this.

Alternatively, Leif shows that for HF you can use a simple function generator generating a square wave.

>   https://www.youtube.com/embed/mY3D8pSWsJQ

I have not tried this.

Setting FFT parameters

Resolution bandwidth.

To see what the current parameters are, from the normal operating screen (with waterfall etc), press X, then P (change Parameters). This will provide a display window, the top line of which gives, in my setup for SSB, "fft1 size=2048 (Bw=93.75 Hz)" and then a list of parameters beginning with "First FFT bandwidth (Hz) [100]". This indicates that I had requested a (Resolution) Band Width (RBW) of 100 Hz, but the actual closest value was 93.75 Hz.

Changing the value of "First FFT bandwidth (Hz)" to 50 on this screen, changes the top line to "fft1 size=4096 (Bw=46.9 Hz)". The resolution of the main waterfall will now be finer if you have sufficient pixels available to display it. The waterfall speed will also decrease.

Note that the maximum resolution you can actually see depends on your display. For example, my cheap laptop screen has a horizontal width of only 768 pixels. The Linrad waterfall occupies 25 mm of the total 35 mm, thus there are only 548 pixels in the waterfall. If I have the waterfall set to display 96 kHz of bandwidth, the maximum resolution is 96000/548 or 175 Hz. For this displayed bandwidth there would be no advantage to reduce the RBW to 50 Hz, even 100 is more than can be displayed. However, if you reduce the displayed bandwidth using the arrows at the top of the waterfall window, then the displayed resolution increases up to the one pixel per RBW point. If you need more resolution, then you need to

decrease the "First FFT bandwidth" value to 50, 20 or whatever you need. Note that waterfall speed will then be slowed.

First mixer bandwidth reduction.

http://www.sm5bsz.com/linuxdsp/usage/newco/newcomer.htm

says re. "First mixer bandwidth reduction" *"The following values can serve as guidelines:*

```
Rate        Hardware     Mix1 bw parameter
(kHz)        Type        CW     SSB     AM
 48         3,4           4      3       1
 96         3,4           5      4       2
192         3,4           6      5       3
```

The Softrock is hardware type 3 and the "Rate" is the soundcard sampling rate.

The default rates that I had accepted originally for my 96000 rate were 4,4,2 for normal cw, ssb, am, so there is probably no need to change these.

Transmit with Softrock RXTX

Leif is working on this. It is only partly implemented in version 04.08.

Links to additional resource material

A guide similar to this one prepared by ON4KHG for an early version of Linrad. This explains features and parameters in more detail. The main page is in French. Scroll down to the red line **Linrad Installation and Configuration User Guide** where there is a download link.

http://on4khg.be/linrad-by-leif-sm5bsz/

Info on the Linrad screen controls (buttons). It is possible that some of this is out of date, I have not checked it over.

http://physics.princeton.edu/pulsar/K1JT/Linrad_On-Screen_Controls.pdf

Info on FFT parameters

http://www.nitehawk.com/linrad_dat/linrad_params.htm

http://www.nitehawk.com/linrad_dat/linrad_params_1.html

This video has Leif showing how to maximize settings to show weak signals. This illustrates adjusting FFT parameters and waterfall parameters. This one is quite helpful for understanding how the parameters affect the display, resolution, waterfall speed.

https://www.youtube.com/watch?v=C-oUcltshz4

This shows the extreme control provided over FFT parameters.

http://g7rau.demon.co.uk/sm5bsz/linuxdsp/install/dlevel.htm

Also this one using Linrad as a radar for meteor shower reflections.

http://www.sm5bsz.com/linuxdsp/usage/radar/pers2008/pers2008.htm

Upgrading Linrad

I upgraded a laptop running 04.07 to 04.08.

When I installed 04.08 it installed in a new "04.08" folder under the 04.07 one. I did not want to have to redo all the setup, I/Q calibration and screen customizations I had done for 04.07.

I copied all the "par_ xxx" and "dsp_xxx" files from the 04.07 folder to the 04.08 folder. When I started 04.08, it still wanted me to do the "Global parameters" setup and to select the soundcards, which takes only a minute, but it accepted all the other setup data.