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Linrad Installation & Configuration User Guide

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Contents

Glossary

A/D : Analogue to Digital (converter)
AFC : Automatic Frequency Control
AGC : Automatic Gain Control
ALSA : Advanced Linux Sound Architecture
ASIO : Audio Stream Input/Output
BFO : Beat Frequency Oscillator
CAT : Computer Aided Transceiver
CPU : Central Processing Unit
D/A : Digital to Analogue (converter)
DOS : Disk Operating System
EME : Earth Moon Earth
FFT : Fast Fourier Transform
GUI : Graphical User Interface
IFFT : Inverse Fast Fourier Transform
IP : Internet Protocol
I/Q : In phase / Quadrature
LAN : Local Area Network
MME : MultiMedia Extensions
MMX : Matrix Math eXtension
Portaudio : PORTable cross-platform AUDIO
SDR : Software Defined Radio
S/N : Signal to Noise ratio
VAC : Virtual Audio Cable

1. Introduction

This document describes the procedure to install and configure “Linrad”, a software package developed by Leif, SM5BSZ and which supports Software Defined Radio (SDR) operation. Though Linrad can work under other operating systems (Linux or MAC), the present document focuses on running Linrad under Microsoft Windows XP, Vista or Win7 operating systems. I’m not “married” with Microsoft but I haven’t had time so far to gain interest and investigate open source operating systems.

Linrad is so much powerful (with features that can’t be found on any other SDR software) and versatile in the way everything can be customized through an exhaustive list of parameters. Hence, at first sight, it could seem complicated to configure but once the “philosophy” behind Linrad has been understood, it is not that complicated. The aim of the present document is to help demystifying Linrad for newcomers (beside the extensive information already available on the website of Leif, http://www.sm5bsz.com). It is not possible to cover here everything (hardware and configuration-wise), given Linrad offers so many possibilities. So, I “only” describe the way I myself use Linrad, associated with a given specific hardware but there are many others ways not exposed in this document.

Two installation possibilities are described here :

- An easy way is to download **wlrx-yy.zip** (x-yy is the software version), a **pre-compiled version** of Linrad and unzip it into a suitable directory

  OR

- You may also decide to **compile the executable** yourself by downloading **lir0x-yy.zip** and extracting the Linrad source code into a suitable directory. This allows for example to add a “CAT window” feature in your Linrad, in order to control the frequency of a transceiver linked to your computer through a COM port (not possible with the pre-compiled version above).

While installing Linrad on a computer (by one of the two means above), nothing is written in the computer registries, so that all the Linrad files can easily be copied to any other directory or deleted, without needing to run “uninstall” or access the computer control panel.

To save your printer ink, most screenshots in the present document are color-reversed (black appears white and vice versa).

One will always refer to the website of Leif for an information up to date.

A wide part of the present document is based on the help file (help.lir) provided in the Linrad package (the help of Linrad version 3.36 is used here). However, having the help printed on a document like this one can sometimes be more convenient, as it can also for example be read off-line.

For an efficient usage of it, Linard requires a minimum deal of personal involvement, though it is not needed to be an expert in Fourier transforms, bins, averaging,… The difficulty encountered by newcomers resides in how to set so many parameters for one’s own usage and RF environment. To help in this purpose, in section 6 of the document, one can find the parameters I use for different modes (SSB, CW, Weak CW for beacon monitoring, FSK441 or JT65), either for Linrad as a stand-alone receiver, as a remote receiver or together with WSJT9 and MAP65(-IQ). I don’t claim to be an expert ; I have spent many hour tweaking the parameters, so that now I have parameter sets meeting my requirements and environment.

Don’t copy them all “blindly” ; many parameters still require some degree of customization to accommodate your own specific hardware and/or environment. Indeed, the position of the windows on your screen depends on your screen resolution (not necessarily the same than mine), the adjustment of levels and gains will depend on your own whole RX RF chain gains, and so on. However, after having read this document in a row or having consulted specific sections, you should be able to do it yourself.

2. Installation

2.1. Prerequisite

Several libraries need to be installed first.

The easiest is to download and execute the installer setup-linrad-dll-package-02.exe to be downloaded at http://www.sm5bsz.com/linuxdsp/install/compile/wincompile.html
It will place the files palir-01.dll, palir-02.dll, libusb0.dll and inpout32.dll in your {sys} directory.

2.2. Pre-compiled version

- Download the file **wlrx-yy.zip** on http://www.sm5bsz.com/linuxdsp/linroot.htm and unzip it in whatever directory you wish.
- Launch **linrad.exe**, you get the following window :

```
WELCOME TO LINRAD
This message is not an error, but an indication that setup
has not yet been done.

Setup file par_userint missing.
Use W to create a new par_userint file after setup.

Note that the following keys have a special meaning in Linrad:
ESC = terminate Linrad
 X  = Skip whatever process you are in and get one level
      upwards in Linrads menu three.
 G  = Make a .gif file with a screen dump of your current screen.


 -----------  GLOBAL PARAMETERS SETUP -------------
     (You might want to edit par_userint instead)
Press N for NEWCOMER mode.
Press S for normal mode.
Press E for expert mode.
Then press enter

=>e_
```

- Press “E” + Enter.
- You are prompted to define the font scale (size of the characters) ➔ type “2” + Enter (you can still change it afterwards without running again the whole installation process).
- Set the process priority (the priority of Linrad amongst the other applications running, if any) ➔ type “1” + Enter.
- Amount of processors (CPU’s) to devote to Linrad. Type “0” + Enter.

```
Enter font scale (1 to 5), then press Enter: 2

Set process priority (0=NORMAL to )3=REALTIME, then press Enter: 1

This system has  2 processors.
How many do you allow Linrad to block?
If you run several instances of Linrad on one multiprocessor
platform it may be a bad idea to allow the total number
of blocked CPUs to be more that the total number less one.

=>0

Set timer resolution in 1 to 999 ms (0 to use default)
=>0

Set autostart: Z=none,A=WCW,B=CW,C=MS,D=SSB,E=FM,F=AM,G=QRSS
=>z

Percentage of screen width to use(25 to 100):
=>100

Percentage of screen height to use(25 to 100):
=>94_
```

- Set the timer resolution to “0” + Enter.

- Set the autostart. This allows to run Linrad directly in the wanted mode (SSB, CW,…) without passing through the main menu. Start with "Z" + Enter (no autostart, it can be changed afterwards).
- Percentage of screen width Linrad will occupy on your screen. I choose "100" + Enter but this is machine dependant ; another value could be more suitable on other types of screens.
- Percentage of screen height. Start with 80 ; you will subsequently increase this figure until you can still see the % of CPU usage at the bottom left of the screen while in RX mode. In my case, I choose "94" + Enter.
- A new window appears. If not done automatically, minimize the preceding one but don't close it.

```
                Linrad-03.33    Soundcard
                expert mode

A=Weak signal CW                  1=Process first file named in 'adfile'
B=Normal CW                       2=Process first file named in 'adwav'
C=Meteor scatter CW               3=Select file from 'adfile'
D=SSB                             4=Select file from 'adwav'
E=FM                              5=File converter .raw to .wav
F=AM                              T=Toggle network output
G=QRSS CW
H=TX TEST
I=SOUNDCARD TEST MODE
J=ANALOG HARDWARE TUNE
K=RADAR

M=Init moon tracking and EME database
N=Network set up
S=Global parms set up
U=A/D and D/A set up for RX
V=TX mode set up
W=Save current parameters in par_userint
F9=Emergency light
F1 or !=Show keyboard commands (HELP)

     PARAMETERS NOT SAVED Press W to save
```

- Type on "W" to save the parameters.
- Start with selecting the sound devices (A/D and D/A set up for RX), type "U".

```
CURRENT A/D and D/A SETUP FOR RX

Linrad RX input from: SOUNDCARD NOT YET SELECTED: Select Menu Option A

Linrad RX output to:  NOT YET SELECTED (SHOULD BE EITHER SOUNDCARD OR DISABLED, AND/OR NETWORK)


DMA rate   min=30    max=300


A = Change the input settings and reset the output soundcard settings if a soundcard is selected as input.
B = Change the output soundcard settings.
C = Change min/max dma rate.
E = Enable/Disable frequency converter and set shift.
Z = Disable the output soundcard.
X = To main menu.
```

- Type "A" to select the input audio device amongst the list.

```
            SELECT HARDWARE FOR INPUT

   A = Soundcard
   B = SDR-14 or SDR-IQ
   C = Perseus
   D = SDR-IP
   E = Excalibur
   F = libExtIO hardwares
   Y = Network
   Z = Disable (Disk input allowed)
```

- It depends on your specific SDR hardware but in my case, I select “A” for Soundcard. Since specific libraries are located in the Linrad directory, other sound devices can also be supported.
- You are prompted to use Portaudio or not. I haven’t been very successful in using it so far. So, I select “N” but feel free to try it. The benefit is that it might allow the usage of more channels and more bits under Vista and Win 7 in case the MME driver does not support WAVEFORMATEXTENSIBLE (see below). MME drivers are slow with Portaudio.

```
Use Portaudio for rx input? (Y/N) =>
```

You get the list of soundcards and you have to select the number in front of the name of the soundcard you want to use (e.g. “5” for the E-MU 0202).

```
Select SOUNDCARD device for RX input from list

0  Realtek AC97 Audio
1  Virtual Cable 1
2  Virtual Cable 2
3  Virtual Cable 3
4  SB Live! 24-bit External
5  E-MU 0202 | USB


Select (first) device for Rx input by line number> _
```

- “Use extended format (WAVEFORMATEXTENSIBLE) ?” If your soundcard supports 24 bits, type on “Y”. If it supports 16 bits only, you can type on “N”. In my case, it is “Y”.
- “Sampling speed (Hz)” ➔ depends on your soundcard. For mine, it is “192000” + Enter. If Linrad is to be used with MAP65, type “96000” anyway + Enter.
- “No of bits (16/24”) ➔ 16 or 24 + Enter (hardware dependant).

```
Select SOUNDCARD device for RX input from list

0  Realtek AC97 Audio
1  Virtual Cable 1
2  Virtual Cable 2
3  Virtual Cable 3
4  SB Live! 24-bit External
5  E-MU 0202 | USB


Select (first) device for Rx input by line number> 5
Do you need more channels from the same soundcard ? (Y/N)
F1 for info/help
Linrad can not query hardware because Windows will report that
everything is possible. Windows will silently resample and provide
data that would be meaningless in an SDR context.
Therefore, make sure you enter data that is compatible with the
native capabilities of your soundcardhardware. (And make sure that
the soundcard really is set to the speed you have selected.)

Use extended format (WAVEFORMATEXTENSIBLE) ? (Y/N)

Sampling speed (Hz)> 192000   No of bits (16/24): 24
```

- You are prompted to select the type of SDR hardware. In most cases, you will select “2” (single polarization RX = 1 RF & 2 audio channels, the I and the Q) or “4” (for 2 RX’s and double polarization).

```
          Select radio interface>

          1: One RF, one audio channel (normal audio)
          2: One RF, two audio channels (direct conversion)
          3: Two RF, two audio channels (normal audio, adaptive polarization)
          4: Two RF, four audio channels (direct conversion, adaptive polarization)

          F1 for help/info


Number of points to time shift between I and Q? (-4 to +4)   0_
```

- In my case, I type on “2” and then “0” for the “Number of points to time shift between I & Q ?” Finally, type Enter.

```
          Select receiver hardware to use with soundcard.


     0    Undefined
     1    WSE
     2    Si570
     3    Soft66

Select by line number=>   0
```

- Through Linrad, you can control the (center) frequency of external devices (amongst others a Si570 DDS). I’m not using this feature, so I type “0” + Enter.
- Now, you have to select which kind of output you wish for Linrad.

```
CURRENT A/D and D/A SETUP FOR RX


Linrad RX input from: SOUNDCARD device    = E-MU 0202 | USB
                      device number       = 5
                      associated radio    = Undefined
                      sample rate         = 192000
                      no of input bytes   = 4 (32 bits)
                      radio interface     = One Rx channel, two audio channels
                                            (direct conversion, time shift=0)


Linrad RX output to:  NOT YET SELECTED (SHOULD BE EITHER SOUNDCARD OR DISABLED, AND/OR NETWORK)



DMA rate   min=30     max=300



A = Change the input settings and reset the output soundcard settings if a soundcard is selected as input.
B = Change the output soundcard settings.
C = Change min/max dma rate.
E = Enable/Disable frequency converter and set shift.
Z = Disable the output soundcard.
X = To main menu.
```

- Type on “B” and repeat the same steps as for the RX input. Sometimes using the same soundcard for input and output can cause problems. If possible, better use another one for output. As an output, sending data streams to a local network is also possible ; we will come back on this later.

```
CURRENT A/D and D/A SETUP FOR RX


Linrad RX input from: SOUNDCARD device    = E-MU 0202 | USB
                      device number       = 5
                      associated radio    = Undefined
                      sample rate         = 192000
                      no of input bytes   = 4 (32 bits)
                      radio interface     = One Rx channel, two audio channels
                                            (direct conversion, time shift=0)


Linrad RX output to:  SOUNDCARD device    = SB Live! 24-bit External
                      device number       = 0
                      hostapi             = Native MME
                      min da sample rate  = 8000
                      max da sample rate  = 96000
                      min da bytes        = 1
                      max da bytes        = 2
                      min da channels     = 1
                      max da channels     = 2



DMA rate   min=30     max=300



A = Change the input settings and reset the output soundcard settings if a soundcard is selected as input.
B = Change the output soundcard settings.
C = Change min/max dma rate.
E = Enable/Disable frequency converter and set shift.
Z = Disable the output soundcard.
X = To main menu.

(Do not forget to save with W on the main menu)
Press any key
```

- Type on “X” + press any key, you are back to main menu.

```
              Linrad-03.33    Soundcard
              expert mode

A=Weak signal CW                1=Process first file named in 'adfile'
B=Normal CW                     2=Process first file named in 'adwav'
C=Meteor scatter CW             3=Select file from 'adfile'
D=SSB                           4=Select file from 'adwav'
E=FM                            5=File converter .raw to .wav
F=AM                            T=Toggle network output
G=QRSS CW
H=TX TEST
I=SOUNDCARD TEST MODE
J=ANALOG HARDWARE TUNE
K=RADAR

M=Init moon tracking and EME database
N=Network set up
S=Global parms set up
U=A/D and D/A set up for RX
V=TX mode set up
W=Save current parameters in par_userint
F9=Emergency light
F1 or !=Show keyboard commands (HELP)

     PARAMETERS NOT SAVED Press W to save
```

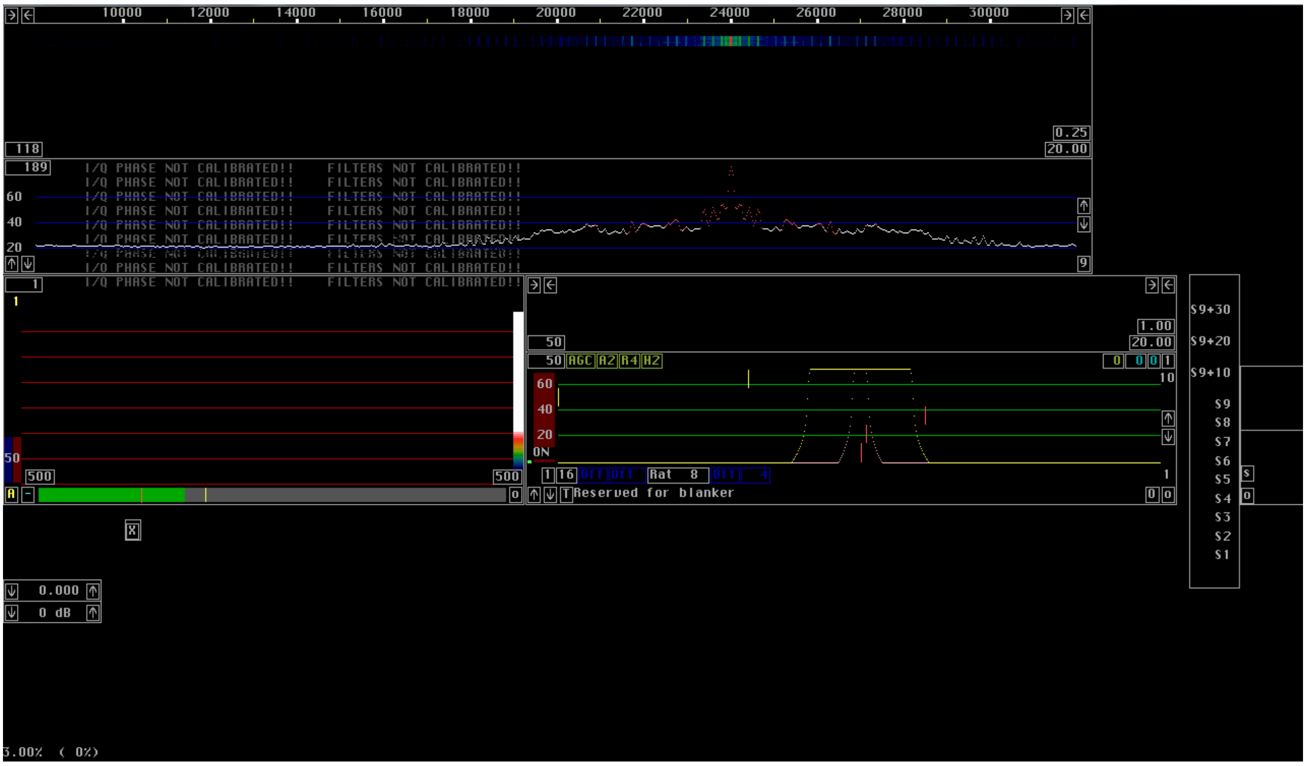
- Type on “W” to save what has been done so far. Now, in the directory where Linrad is installed, an additional file has been created, par_userint. This file contains **machine related parameters** (screen, soundboard,...) defined up to now. These can be subsequently changed through the menus “S” (Global parms set up) and “U” (A/D and D/A set up for RX).
- Now, **mode related parameters** need to be defined. Type on “D” for SSB related parameters.

```
You are prompted to the parameter selection screens

           for the following reason:


par_ssb file missing


                 Press any key
```

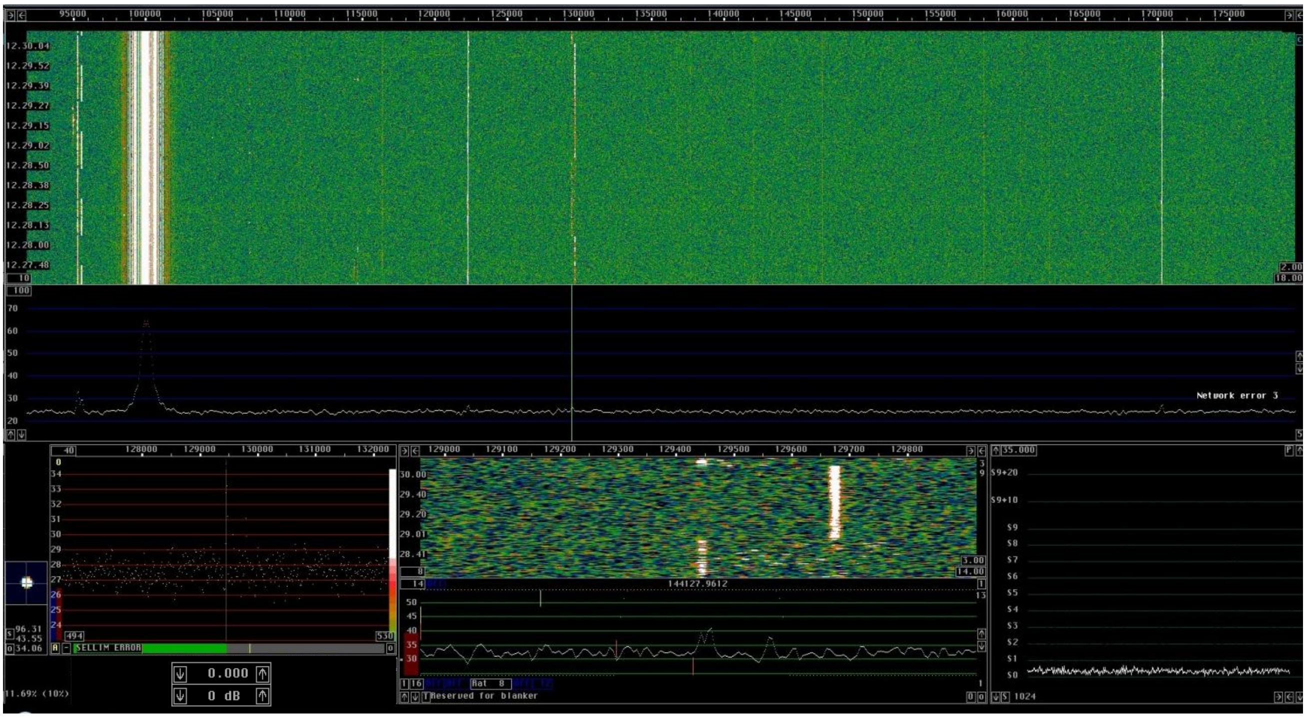
- Press any key.

```
     SSB: Rx channels=1    fft1 size=4096 (Bw=93.750000Hz)

First FFT bandwidth (Hz) [100]
First FFT window (power of sin) [2]
First forward FFT version [0]
First FFT storage time (s) [1]
First FFT amplitude [2000]
Main waterfall saturate limit [0]
Enable correlation spetrum [0]
Enable second FFT [0]
CONTINUE

Use left mouse button to select line
```

- Mouse left click on the values between [ ] to change them. When done, type “Enter”. And so on for every parameter you want to change. The goal is not at this stage to discuss and enter every single parameter in detail. Click on “Continue” to move to the next screen, up to when you get something like the following screen.



- You can move the windows around to make the screen suit your taste. It has to be done by moving border lines and sometimes one has to move the other side first because the window size might be restricted. It may also be necessary to move another window first because windows avoid each other. You may come up with something like the following screenshot.



Additional information :

1) Once you change something (e.g. the size of a window or a parameter on the graphical user interface), the change is stored in the parameters and will still be effective at next start up of Linrad. It is wise to create a directory with an instance of Linrad that could be used for trial/learning

purpose, onto which trials around the parameters could be performed. So, the main Linrad instance used as RX stays untouched, until a satisfactory change has been established on the trial/learning instance and translated to the main Linrad.
After the trial/learning directory has been created, just copy all the files already present in the main Linrad directory to the new trial/learning directory. In both directories, you have a file named "linrad.exe". It is better to rename them (e.g. Main Linrad V3-36.exe & Trial Linard V3-36.exe) and create associated shortcuts on your desktop.

2) **Addition of the "EME window"**, which provides the Azimuth and Elevation of the moon, as well as the locator of a distant station since it is included in a known list, the "call3.txt" file. This file contains thousands of callsigns and their associated locator ; it is provided together with WSJT. It can also be downloaded on Make More Miles on VHF (file managed by Guido, DL8EBW). To get the EME window when opening Linrad, proceed as follows :

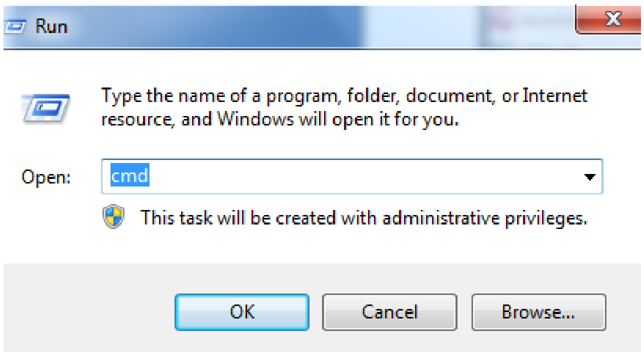
- Create a directory C:\emedir (or a dir. named "emedir" in the dir. where Linrad is installed).
- Place the file "call3.txt" in this directory. As far as I remember, there is a format mistake in the file that prevented Linrad to work properly. I found and corrected the mistake but I don't recall exactly the correction I made (it was very minor). Though it is not up to date (dating summer 2010), you can use my corrected file, by downloading it from my website http://www.on4khg.be/sdr.
- Launch Linrad.
- Type "M" (Init moon tracking and EME database) in the main menu.
- Enter your own QRA Locator.
- Set "Auto Init" to "1".
- Type "9" to save and come back to the main menu.
- Start the mode you wish (e.g. : "D" for SSB) and very small window with a "X" will appear somewhere on the screen ➔ 
- Click on the "X" and the EME window will pop up. As the Auto Init has been set to 1, this same window will appear again at the subsequent launches of Linrad.

2.3. Compile the executable

If you compile yourself the executable file, you will be able to get the CAT window which allows to set an analogue transceiver on the same frequency than Linrad, through COM (computer) and CAT (transceiver) ports. CAT stands for "Computer Aided Transceiver". Thanks to Pierre, ON5GN for development & support.

You just have to press "Q" on the computer keyboard and the transceiver frequency moves to the one onto Linrad is set to. This is particularly interesting, since Linrad has no TX capability so far. To get this window, proceed as follows :

- Download the file mingw516nasmusb.zip. (http://www.sm5bsz.com/linuxdsp/install/pa/pa.htm)
- Unzip this file to C:\ ; you will get a subdirectory named C:\MinGW (don't use a different subdirectory !).
- Download Linrad, e.g. **lir0x-yy.zip** (not wlrx-yy.zip) on the Leif's site. (http://www.sm5bsz.com/linuxdsp/linroot.htm). Unzip it in whatever directory.
- In the directory where Linrad is installed, open (with the notepad) the file "users_tr.c" and modify it to suit your needs. Since I use a transceiver (FT-857) which is already embedded in that file (see section 3.5.10. for the list of supported transceivers), no need to modify anything in my case. Save this file with "wusers_hwaredriver.c" as name (in the same directory as the one where Linrad is installed).
- Copy the palir-02.dll file from your {sys} directory to the one where Linrad is installed.
- Open the DOS window (Run/cmd/cd...etc up to the directory where Linrad is installed) and run "configure.exe". The sequence is described below.




```
Microsoft Windows [Version 6.1.7601]
Copyright (c) 2009 Microsoft Corporation.  All rights reserved.

C:\Users\gho>cd..

C:\Users>cd..

C:\>cd program files

C:\Program Files>cd Linrad - ssb - 3.33

C:\Program Files\LINRAD - SSB - 3.33>configure.exe_
```

- After, run “make.bat”. Linrad is now compiled and a file “linrad.exe” is now available in the same directory. You have many more files than when using the pre-compiled version but the “par_...” files are present here too, and identical to the ones installed using the pre-compiled version.
- Run “linrad.exe” and proceed through the menus the same way as in section 2.2. At the end of the process, the CAT window is available. Select the required COM port (the one linking the computer to the transceiver CAT port) and your type of transceiver. You can now control the frequency of your transceiver by pressing “Q” on the computer keyboard.

3. Configuration

3.1. Introduction

In Linrad, there are four types of parameters, embedded in several “par_...” files :

- **Machine** related parameters (computer hardware, soundcards,…).
- **Mode** dependant data processing parameters (the process of data between the input and the output, according to the mode, SSB, CW,…). These parameters are static ; they can be changed only by exiting the RX (press "X" then "P") and setting them in the menu (or with a text editor).
- **User preference** defined parameters (according to the preference of the operator : size of the windows, AGC activated or not,…). These can be changed on the fly by clicking boxes or typing in values in the corresponding Linrad window. These are also mode dependant.
- Here and there in the above mentioned files, there are also dynamic parameters, derived by Linrad itself.

The “par_...” files are all located in the directory where Linrad is installed and, beside configuration through menus, they can also easily be edited with a text editor (notepad). Below in section 3.3., there is the list of the files I have on my system ; the list is not exhaustive, there can be other systems using other files not listed here (e.g. libExt IO.dll to support a specific hardware).

3.2. Linrad general keyboard commands

A = Numerical display, amplitudes.
E = Eliminate spur (if AFC is present).
F = File playback control. Select the part(s) to play.
C = Clear spur elimination tables (if AFC is present).
G = Save entire screen as .gif file. A directory “Linrad_data” must be created (C:\Linrad_data or in the directory where Linrad is installed) ; it is the path where the screenshots will be saved.
M = Toggle freq. stepping mode : Center, Passband or BFO (see also Left, Right).
S = Save input data as file (S again to stop).
T = Numerical display, processing delays.
W = Write output data as .wav file (W again to stop). A directory “Linrad_data” must be created (C:\Linrad_data or in the directory where Linrad is installed) ; it is the path where the audio files will be saved.
X = Exit from current mode to menu.
Z = Restart averaging for S-meter and numerical displays.
SPACE BAR = Set the transmitter to the current receive frequency.
V or B or N = Set the receiver to the current transmit frequency.
F1 or ! = Help. (place mouse on a screen object).
F1 or ! = Show screen objects (place mouse outside windows on black screen).
(F1 or ! once more = Show thread id. Only under modern Linux).
F2 = Toggle S-meter averaging.
F3 = Skip D/A output (for fast processing of disk files.)
Shift F3 = Stop disk file processing (to study morse decode intermediates).
F4 = Show RF envelope vs time as an oscilloscope function in TX test mode.
F5 = Pause processing. Useful for wideband recordings on slow computers.
F8 = Reduce the strong signal when F11 is in use.
F9 = Truncate input (for test only).
F10 = Add noise when F11 is in use (Linux only).
F11 = Enable internal signal generator.
Up = Increase bandwidth.

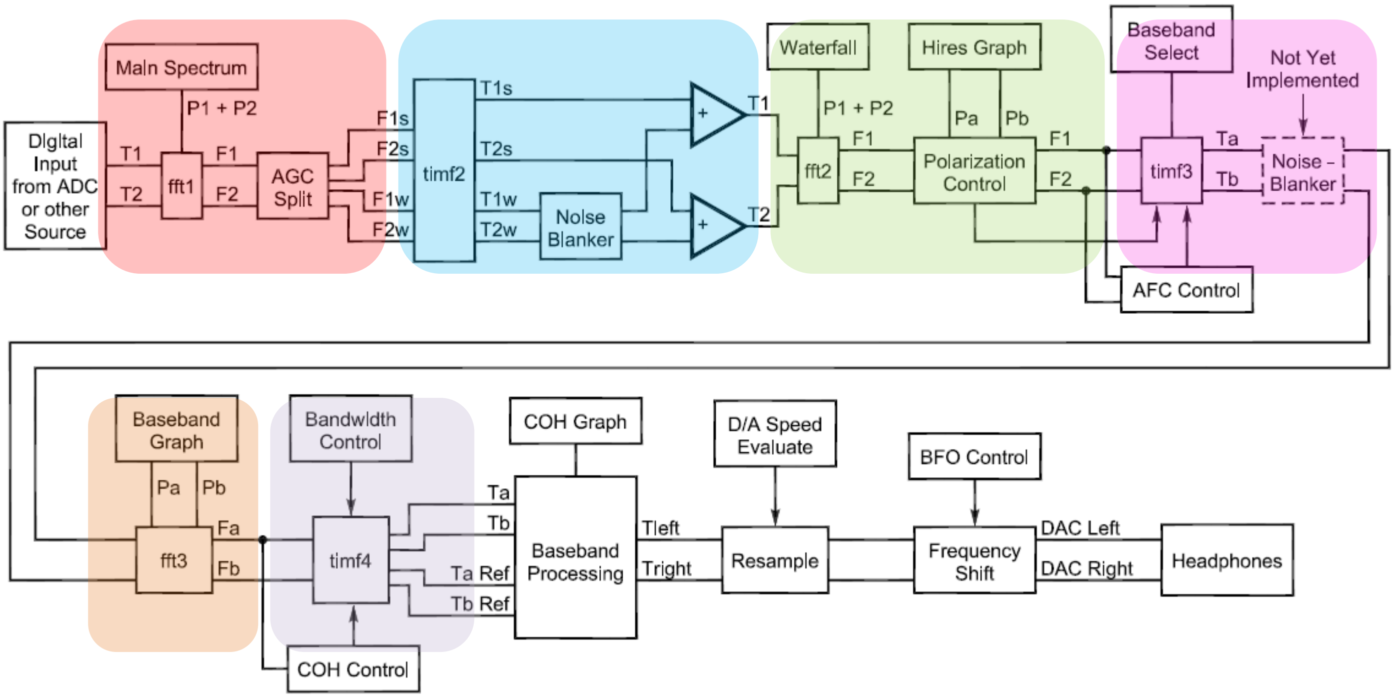
Down = Decrease bandwidth.
Left = Decrease frequency (“M” or mouse box for mode).
Right = Increase frequency (“M” or mouse box for mode).
+ = Make the wheel mouse step size twice as large.
- = Make the wheel mouse step size twice as small.
(Wheel is equivalent to Left and Right. Pressed to “+” or “-”)
ESC = Quit from program

3.3. Summary of Linrad parameter files

Parameter file	Type of embedded parameters	Configured through menu(s)	For more details, see section(s)
Par_userint	Machine	• S (Global parms set up) • U (A/D and D/A set up for RX)	2.2. & 2.3.
Par_network Par_netsend_ip	Machine	• N (Network set up)	4.2.2.1., 4.2.2.2., 4.2.3.1., 4.2.4.1., 4.2.5. & 4.2.6.
Par_xxx xxx = ssb, wcw, am,…	Mode	• D (SSB) • A (WCW, Weak CW) • …	3.4.2.
Par_xxx_bg bg = baseband graph	User Preference	No configuration via menus but on the fly, via the graphical user interface (the Linrad windows) OR directly in the file, via a text editor	3.5.1.
Par_xxx_cg cg = coherence graph			3.5.2.
Par_xxx_eg eg = EME graph			3.5.3. & end of 2.2.
Par_xxx_fg fg = frequency graph			3.5.4.
Par_xxx_hg hg = high-resolution graph			3.5.5.
Par_xxx_mg mg = meter graph			3.5.6.
Par_xxx_wg wg = waterfall graph			3.5.7.
Par_xxx_pg pg = polarization graph			3.5.8.
Par_yyy_ag (yyy = wcw or cw) ag = AFC graph	User Preference, only available in (W)CW mode		3.5.9.
Par_xxx_hwd_ug hwd = hardware driver ug = user graph	User Preference (for the CAT window, after compilation of the executable)		3.5.10.

3.4. Mode dependant parameters

3.4.1. Block architecture of Linrad



The **FFT** stages are “Fast Fourier Transform” stages ; these “convert” a signal varying in the time domain (T) as can be seen on an oscilloscope into a one varying in the frequency domain (F), as seen on a spectrum analyser. The reverse or backward process (from frequency domain to time domain), is denoted **timf** on the block diagram.

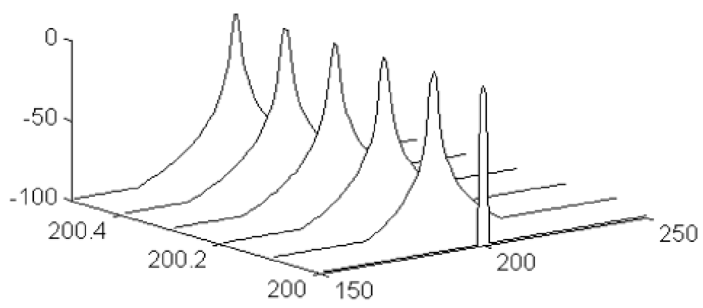
Block	Function	Associated parameters in the “par_xxx” file (xxx = ssb, cw, am,…)
	FFT1	• First FFT bandwidth (Hz) • First FFT window (power of sin) • First forward FFT version • First FFT storage time (s) • First FFT amplitude
	timf2 (= IFFT1, inverse or backward FFT1)	• First backward FFT version • First backward FFT att. N
	FFT2	• Enable second FFT • Second FFT bandwidth factor in powers of 2 • Second FFT window (power of sin) • Second forward FFT version • Second forward FFT att. N • Second FFT storage time
	timf3 (= IFFT2)	
	FFT3	• Third FFT window (power of sin)
	timf4 (= IFFT3)	

3.4.2. List of mode dependant parameters (par_xxx)

The parameters are listed according to the ranking as they appear in the Linrad GUI ; they are highlighted in different colors, each corresponding to a block of parameters, still as in the GUI.

First FFT bandwidth (Hz) [0-6000]
The first FFT bandwidth is used to set the size of the first FFT to a bandwidth within a factor of two of the desired value. If the second FFT is enabled, there is no reason to select a very narrow bandwidth since the first FFT then is used only for the selective limiter that prevents strong signals from reaching the noise blanker. When second FFT is disabled the first FFT bandwidth will directly affect the sensitivity of the waterfall graph and it will also affect AFC performance. Narrow bandwidths cause appreciable processing delay, this is unavoidable since data has to be collected over a time period that is about twice the time given by 1/bandwidth, (depends on what window is selected).
For a sampling rate of 96 kHz and a FFT size of 1024 points, the resolution bandwidth equals 96000/1024 = 94 Hz if no windowing is applied.

First FFT window (power of sin) [0-9]
The problem of broadening of signals due to mismatch at the ends is solved by windowing. The trick is simple: Force the points to zero at both ends by multiplication by some function that gradually goes to zero at both ends. Obviously a string of zeroes at one end will give a perfect match with the string of zeroes at the other. Windowing removes the tails that stretch far away from the centre frequency, and thus it gives a very significant improvement of the dynamic range. When windowing is used, the resolution becomes lower. The frequency response becomes broader also for sine waves that fit exactly to the Fourier frequencies. On the other hand the broadening is controlled and independent on whether the signal is right at a Fourier frequency or somewhere between.



The window function used for the first FFT is a power of sin(x) from 0 to 7. The figure on the left shows a sine squared window ➔ "First FFT window (power of sin)" = 2. The parameter value 8 gives a Gaussian window, and the parameter value 9 generates a flat window that falls off to (near) zero with the Gaussian error function erf(x). For a short delay from antenna to loudspeaker, select 9 and a bandwidth of 300 Hz. For minimum CPU load (and somewhat compromised dynamic range), select 1 and a bandwidth of 10 Hz. For precise spectral analysis, select 7 or 8 and whatever bandwidth you like to see on screen.

First forward FFT version [0-2]
Version number for FFT1 routine. Depending on your hardware there are several different FFT implementations available. Check the timing of each one of them and select the fastest if there is a significant difference.

First FFT storage time (s) [0-2000]
This parameter affects memory allocation for old FFT1 transforms. It affects the maximum number of averages you may use for the main spectrum. In case second FFT is deselected this parameter also affects the memory allocation for AFC and spur rejection in case these functions are enabled.

First FFT amplitude [1-1000000]
If you have disabled the second FFT, this parameter will just shift the dB scales just like a volume control. In case you use the second FFT you should use this parameter to set the noise floor at the input to the first backwards FFT. Press A on the keyboard while your system is running to get amplitude information in the lower left corner. "Floor" is the number of bits RMS for the noise floor of the signal entering the first

backwards FFT. A larger value for first FFT amplitude will increase “Floor”. Your loss of system noise figure because of quantization noise will be:

Floor (RMS voltage)	NF loss (dB)
1	0.4
2	0.2
4	0.1
10	0.04

Placing the noise floor too high may lead to saturation in later processing steps. On good hardware with soundcards, the default parameter value 1000 should give a floor value of about 10 when the hardware is running at full gain with an antenna connected. The system noise floor should then be about 20 dB above the noise of the soundcard alone. In case you need a smaller value than 1000 for this parameter, your system is likely not optimised for dynamic range. Too much analogue hardware gain. In case you need a larger value than 1000 your system is likely insensitive, a preamplifier would likely improve the system NF. When evaluating .wav files, this parameter may have to be set as low as 1.

Main waterfall saturate limit [0-99]
Use this parameter to make the main waterfall black during transmit in case your transmit signal gets into the receiver and produces a saturated spectrum. The waterfall gain and zero level determines what signal level is required to saturate. This parameter is the percentage of points that may saturate (and become white) without causing the waterfall line to become blanked out (black). Setting this parameter zero will disable the waterfall blanking.

Enable correlation spetrum [0-1]
Enable the correlation spectrum between two RF channels to be displayed in the main spectrum window. This function can be used for sideband noise measurements when the two RX channels have uncorrelated sideband noise. Average 200 times for a 10dB noise floor improvement or 20000 times for a 20 dB improvement. The factor 2 from conventional instruments is due to the overlapping transforms in Linrad when using sine squared windows ; other windows have slightly different overlapping. Use a wide bandwidth for the averaging to not take too much time.

Enable second FFT [0-1]
The second FFT is the high resolution FFT that is intended for use at large bandwidths (20kHz analogue bandwidth and more). If your analogue hardware is a conventional radio with a few kHz bandwidth there is usually no reason to enable the second FFT. Only if your radio has very low distortion for signals within the passband and you are troubled by impulse noise that can not be removed by the noise blanker of your radio because of the presence of strong signals at nearby frequencies ; the second FFT will be useful since it allows noise blanking in the presence of strong signals. Note that the second FFT has to be enabled for the Linrad noise blankers to become available. To use the smart blanker your system also has to be calibrated.

First backward FFT version [0-1]
Version number for back transformation from FFT1 to timf2. Select 0 for floating point, 1 for 16 bit MMX (only very old computers do not have MMX). The MMX implementation of the backwards FFT runs much faster than the floating point implementation but it requires a little more care in setting other parameters on this parameter screen. Note also that the 16 bit MMX routine looses one bit if the window for the first FFT is not 0 or 2. Use a sine squared window for the first FFT with the 16 bit MMX routine.

Sellim maxlevel [200-1000000]
This parameter is needed when 16 bit MMX implementations are used for backwards FFT1 and/or for FFT2. It is used for the frequency selective AGC and sets the largest amplitude allowed on spectral lines at the input of the first FFT backwards transformation. A lower value here will allow more gain in the second FFT

without risk of overflows in 16 bit MMX computations. Press A on the keyboard while your system is running to get amplitude information in the lower left corner. If “fft1 St”, “timf2 St” or “fft2” overflows (margin becomes zero) you may try to decrease this parameter. A small value here may make desired signals distorted because they are classified as “strong” and become compressed in a way that degrades S/N. In case floating point math is selected the amplitude information will not give the above saturation margins, only “A/D” which tells how many dB are left until your hardware saturates (important enough, but not related to Sellim maxlevel).

First backward FFT att. N [2-6]
Number of zero gain butterfly loops for the first backwards FFT from FFT1 to timf2. Each zero gain loop attenuates by 6 dB so this parameter sets the gain of the first backward FFT processing step. The output, timf2, is the input to the noise blanker. This parameter is important when 16 bit MMX arithmetics is used. Press A on the keyboard while your system is running to get amplitude information in the lower left corner. Make the “First backward FFT att. N” as small as possible, but make sure the amplitude margin of timf2 Wk and Timf2 St does not become zero. Occasional saturation of timf2 St is ok but timf2 Wk should never saturate. Read about “set digital signal levels correctly” on the Linrad Home Page on the Internet.

Second FFT bandwidth factor in powers of 2 [0-10]
Set the bandwidth of the second FFT. The second FFT size can be set equal to or larger than the first FFT. Very narrow bandwidths cause some processing delay but give enhanced sensitivity for very weak CW signals. Narrow bandwidths are required for the AFC to lock properly to extremely weak signals.

Second FFT window (power of sin) [0-4]
The window function used for the second FFT is a power of sin(t). Low powers give faster processing but the filter function associated with each frequency bin gets less steep skirts. When a large bandwidth is selected for FFT2, higher window powers are required to produce enough attenuation a few hundred Hz away from very strong signals. Since very strong signals are already removed by the selective limiter associated with the first FFT there is no reason to use powers above 2. At narrow bandwidths, a few Hz and below it is perfectly ok to skip the window completely and use 0 for this parameter to save processing time.

Second forward FFT version [0-2]
Version number for the second FFT. Values 0 and 1 use float arithmetics. The value 2 selects 16 bit integer arithmetics using the MMX extension. The 16 bit MMX version is about 3 times faster than the float versions and might be needed at high sampling rates. Setting the level parameters correctly is however more critical when 16 bit arithmetics is used. Normally the versions for the first backwards FFT and the second FFT should both be set for the same type of arithmetics. Running one as float and the other as 16 bit integers does not give much advantage over running both as integers. Running both as float is however more forgiving and does allow a slightly better performance in some cases.

Second forward FFT att. N [2-variable]
Number of zero gain butterfly loops for the second FFT. What value to choose depends on your hardware, the FFT1 and the FFT2 parameters you have selected. Press A on the keyboard while your system is running to get amplitude information in the lower left corner. Make the “Second forward FFT att. N” as small as possible, but make sure the amplitude margin of FFT2 does never become zero. Read about “set digital signal levels correctly” on the Linrad Home Page on the Internet.

Second FFT storage time (s) [0-2000]
This parameter reserves memory for storing old FFT2 transforms. Old transforms are used by the AFC. For tracking really weak signals you may need 30 seconds here. Old transforms are also used to recalculate average FFT2 spectra when the frequency is changed. If you can not set large enough numbers for FFT2 avgnum, make this parameter larger. Without AFC a typical FFT2 storage time is 4 seconds.

Enable AFC/SPUR/DECODE (2=auto spur) [0-2]
The AFC and spur removal are sophisticated functions that you can read about on the Linrad Home Page. DECODE is development work on weak signal MORSE decoding. It will probably never be completed. If you enable the AFC or spur removal, make sure you have allowed enough storage for old transforms, “First FFT storage time” or “Second FFT storage time”. 0 = Disable AFC/SPUR/MORSE / 1 = Allow AFC/SPUR/MORSE (manual spur control) / 2 = Allow AFC/SPUR/MORSE (automatic spur control).

AFC lock range Hz [1-800]
Max AFC search range. This parameter and the AFC max drift parameter determine how much memory is allocated for the AFC routine. In case you select a very narrow bandwidth for the FFT's used for the AFC in combination with a very large search range the time to lock to a signal may become excessive. Do not make the search range much larger than required to accommodate for the uncertainty in placing the mouse on a signal in the graphs. If very high resolution is selected for the second FFT, use the high resolution graph to lock to the desired signal.

AFC max drift Hz/minute [0-5000]
Max frequency drift for signal search. The AFC searches for signals that drift linearly with time. Averages of all the old transforms are made under assumption of all possible frequency drifts within the limits given by this parameter. This parameter in combination with the search range parameter determines memory allocation and time consumption of the search process.

Enable Morse decoding [0-1]
Feature of displaying the CW decoding not implemented ; leave this parameter to 0.

Max no of spurs to cancel [0-9362]
Spur removal is a PLL loop that can lock on very stable signals and create a carrier in the opposite phase with the same amplitude. That carrier is then subtracted to remove the spur. The effect is a very deep and very narrow notch filter. The process is efficient in terms of CPU usage, so many spurs can be treated simultaneously. Spur removal operates in the frequency domain using the second FFT if available, otherwise it uses the first FFT. The notch will always be narrower than the FFT resolution. If your spurs are somewhat unstable, increasing the FFT bandwidth may help.

Spur timeconstant (0.1sek) [1-100]
The spur time constant is used to determine from how many transforms the amplitude and phase of the spur should be calculated. The FFT2 storage time (or FFT1 if FFT2 is not present) has to be long enough to keep the transforms in memory.

First mixer bandwidth reduction in powers of 2 [0-10]
The first FFT and the second FFT (if enabled) produce data at the data rate of the A/D board. When a frequency is selected the signal is mixed with an internal digital oscillator of the same frequency to shift the desired signal to zero frequency (baseband I and Q). The signal is then low pass filtered to remove everything above some threshold frequency. When no signal is present above the threshold frequency it is possible to reduce the data rate (sampling speed) without introduction of alias spurs. This parameter determines the maximum bandwidth available for the baseband processing. It also sets the sampling speed for the corresponding time function timf3. The baseband noise limiter (not yet implemented) operates on the timf3 without any selective limiter so it will work only if the baseband bandwidth is set small enough to exclude strong undesired signals.

First mixer no of channels [1-8]
It is possible to select secondary channels by use of the right mouse button. These channels are intended to be processed in the same way as the main channel but they will not be routed to the loudspeaker. The secondary channels are intended to be useful when the automatic implementation of morse code to text is in place. The idea is to allow text information on screen to keep track of what other stations work. The

automatic cw to text translation is not expected to be as good as a trained human operator but it will surely be good enough to allow a good overview of what other stations work.

Third FFT window (power of sin) [1-9]
The window function used for the third FFT is a power of sin(x) from 0 to 7. The parameter value 8 gives a Gaussian window, and the parameter value 9 generates a flat window that falls off to (near) zero with the Gaussian error function erf(x). The window affects the shape of the baseband filter through the different frequency response for each frequency bin that the different windows generate. Linrad versions prior to 02-59 used the value 2 which is generally useful. In case you need steeper skirts with a small time delay, go for a higher number, but not 9. Using 9 here gives a filter with very high stop band attenuation but limited skirt steepness. It provides a very narrow peak and may be useful for coherent processing of AM signals.

Baseband storage time (s) [2-2000]
Time span over which the filtered baseband signal is saved. This is the time over which coherent averaging will search for matching Morse code sequences (not implemented).

Output delay margin (ms) [0-10000]
Set this parameter to a small value to get a small processing delay. Press T to get timing information while receiving. If the D/A MIN value becomes zero the selected value is too small.

Output sampling speed (Hz) [8000-96000]
Some soundcards do not allow different sampling speeds for A/D and D/A. In case very high A/D bandwidths are used, Linrad may need two soundcards to allow a reasonable sampling rate for the output. For morse code there is never any reason to set the output speed above 5kHz. For voice modes in amateur radio it may be useful to set sampling speeds up to 10kHz. The output sampling speed is used also for the generation of .wav audio files and for sending I/Q data over the network.

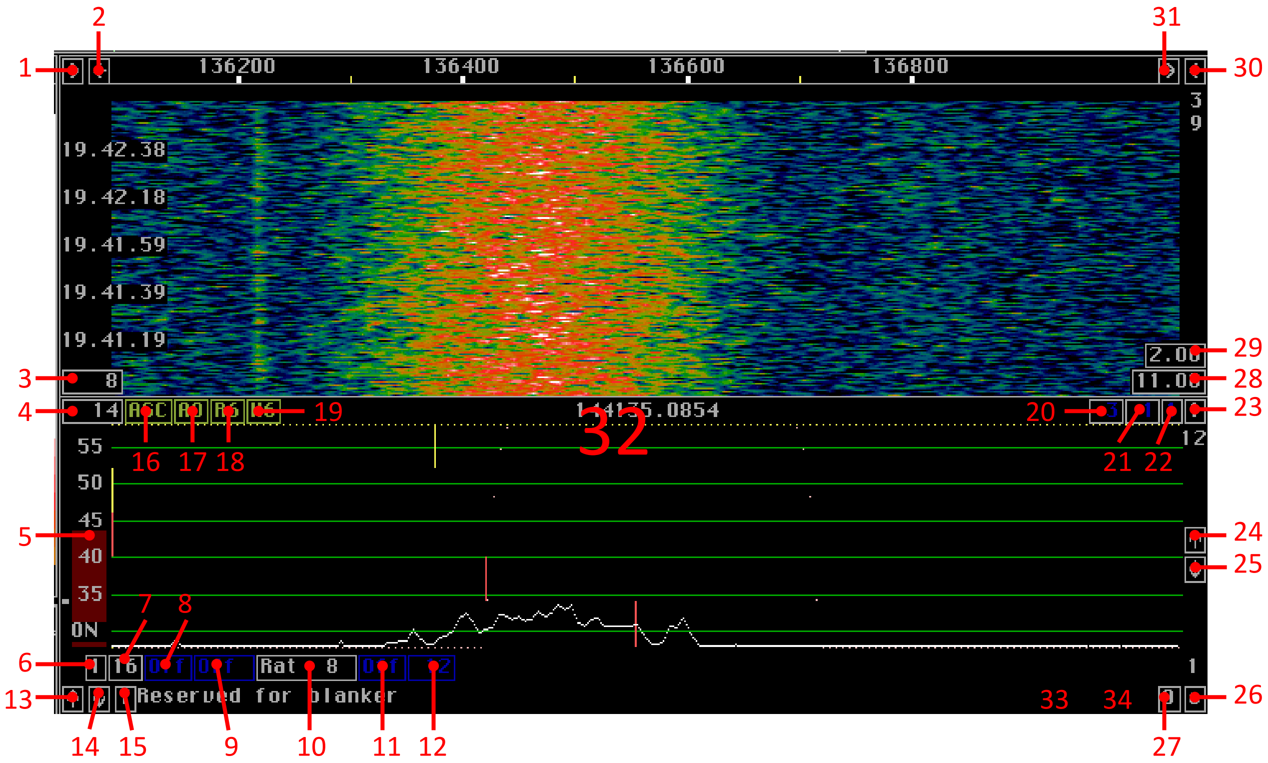
Default output mode [0-200]
The default output mode sets the combination of selections in the line of boxes in the baseband graph to their standard value. The user may select a combination of mode, bits, channels, delay X+Y and other things as the processing for a particular mode (e.g. Coh2 for weak signal CW and X+Y for normal CW). The combination of selections will produce a number (0 to 200) which is displayed at the right hand side of the line of boxes in the baseband graph. Set default output mode to the number you find in the baseband graph when you have selected your favorite processing method for the current receive mode. That way Linrad will always start with the processing method of your choice.

Audio expander exponent [0-9]
The amplitude expander exponent is used to determine the curvature of the expander when the “Exp” option is selected in the baseband graph.

Baseband waterfall saturate limit [0-99]
Same as “Main waterfall saturate limit” but for the baseband window.

3.5. User preference parameters

3.5.1. Baseband window (par_xxx_bg)



1 : Press the left mouse button to **contract the frequency scale**. Resolution will remain unchanged and the amount of interpolation between FFT bins will be reduced.

2 : Press the left mouse button to **expand the frequency scale**. Resolution will remain unchanged and the spectrum is drawn as a straight line interpolation between FFT bins.

3 : Parameter "**waterfall avgnum**". Press left mouse button and type in new value. This parameter controls the averaging for each line of the waterfall graph. Larger values give higher sensitivity but slower response.

4: Parameter "**fft3 avgnum**". Press left mouse button and type in new value. This parameter controls the averaging of the baseband spectrum, FFT3, which has green lines for the dB scale. Larger values give higher sensitivity but slower response.

5 : **Volume control bar**. Click the left mouse button in the bar control area. Note that this volume control is different from the volume control on your loudspeaker or volume controls of the soundboard output mixer. Linrad in weak cw mode is designed to run without any AGC. Strong signals are amplitude limited. Bring this control to maximum to get a saturated signal. Adjust your hardware (loudspeaker and/or soundboard output volumes) for a loud but acceptable maximum sound level. This will be the sound level for saturating signals. When listening to very weak signals, place the volume control bar for the amplitude indicator to the left of it to be placed between 25% and 50% of full scale. The amplitude indicator turns red at the point of saturation and green if the signal is zero (or close to).

6 : Click the left button to force two **output channels**. Can be used only when the mode does not require two output channels. With two output channels for modes where one channel is sufficient the user can select whether he wants the two channels in phase or out of phase. The box immediately left to this one will show "+" or "-" to indicate the sign. In SSB mode "c" will indicate complex format.

7 : Press left mouse button to **toggle output between 8 and 16 bits**. Not really useful - if your hardware allows 16 bit, use them all. With 8 bit you would however get .wav files of half the size and 8 bit data is enough for weak narrowband signals. To get the current mode as default, make the mode parameter “Default output mode” equal to the number at the right hand side of this line of boxes.

8 : Mode for **audio compression/expansion**. “Off” = Normal mode / “Exp” = Amplitude expander on / “Lim” = Simple amplitude clipping.

In normal mode the output is identical to the input up to the saturation level. An extremely fast AGC prevents the input to reach above the saturation level which leads to a limitation of the amplitude at the saturation level without the generation of overtones. When the amplitude expander is on, the amplitude variations of the input signal are expanded by an exponential function. The amplitude expander is intended to be used only at bandwidths below 25Hz where the human ear can no longer hear frequency differences causing both signal and noise to be perceived as a tone with constant frequency and varying amplitude. The human hearing is not designed to distinguish well between small amplitude variations of a single tone because of the logarithmic nature of the amplitude response. The exponential expansion compensates for the logarithmic response and makes decoding of extremely weak signals by ear possible for long times without fatigue. The mode dependant parameter “Audio expander exponent” can be used to set the curvature of the exponential expansion. The simple amplitude clipping is completely linear up to the saturation level where the signal is limited. Overtones are generated when clipping occurs. The onset of overtone generation can be used to hear small amplitude variations for example when listening to aurora signals. When a signal is much wider than the modulation, the optimum filter should match the signal spectrum. The sound character that comes out does not change between key up and key down since it will be the noise spectrum that fits the filter in both cases. The simple amplitude clipping is an alternative to exponential expansion in such cases. To get the current mode as default, make the mode parameter “Default output mode” equal to the number at the right hand side of this line of boxes.

9 : Mode for **coherent processing**. Use left mouse button to select mode: “Off “ = No coherent processing. / “Coh1” = “Binaural CW” / “Coh2” = Coherent : I and Q to left and right ear respectively / “Coh3” = Full coherent processing with I to both ears.

The coherent modes use a second filter that is “Rat” times narrower than the filter shown in yellow on the graph. In binaural CW mode, the narrow filter is typically adapted to the bandwidth of the keying sidebands while the wide filter is 2 to 4 times larger. In coherent modes, the wide filter is adapted to the bandwidth of the keying sidebands while the narrow filter is at least 8 times narrower. (Use button in upper right corner to get more data points on the filter function to allow larger “Rat” values) The Coh2 mode is safe, if the narrow filter fails to follow the carrier due to AFC problems caused by unstable signals or the requirement for small processing delay, no signal at all is lost. If the carrier phase is incorrect, the signal will be present in both ears. This mode is particularly useful for chirping signals. The Coh3 mode is the most sensitive processing mode. It should be combined with AFC with parameters that allow a good locking to the CW carrier. The Coh3 mode gains 3dB S/N by rejecting the noise power that is in the Q channel. There is a further S/N improvement because anything that is in opposite phase to the carrier is also rejected. The Coh3 mode is best used together with “Exp”, exponential expansion of the audio amplitude. Particularly when exponential expansion is used it is important to select the wider filter for minimum possible bandwidth. The narrower filter which is controlled by the Rat parameter must be wide enough to allow the full bandwidth of the carrier to pass. To get the current mode as default, make the mode parameter “Default output mode” equal to the number at the right hand side of this line of boxes.

10 : Press left button to change “Rat”. Enter new value from keyboard. Rat is **the ratio of the bandwidths for the two filters used in coherent modes**. To get the current mode as default, make the mode parameter “Default output mode” equal to the number at the right hand side of this line of boxes.

11 : Press the left mouse button to **toggle delay between ears**. This mode can be enabled only when “Coh” or “X+Y” is not selected. This mode will delay the signal to one ear, thus creating a frequency depending

phase shift between the two ears. The effect of the phase shift is an artificial stereo effect. Different frequencies seem to arrive from different spatial directions. The “Del” mode may be useful when the selected bandwidth is larger than the bandwidth of the desired signal which can be useful for unstable signals or when a very small processing delay not allowing AFC is required. To get the current mode as default, make the mode parameter “Default output mode” equal to the number at the right hand side of this line of boxes.

12 : Press left button to change **the delay between ears** in “Del” mode. Enter new value from keyboard. To get the current mode as default, make the mode parameter “Default output mode” equal to the number at the right hand side of this line of boxes.

13 : Press the left mouse button to **expand the dB scale**.

14 : Press the left mouse button to **contract the dB scale**.

15 : This box shows the current **tuning mode**. Tuning can be made either with a wheel mouse or with the arrow keys. Use “+” or “-” to set the tuning step size. Click the box or press “M” to select one of these modes : T = Tune center frequency / P = Tune passband (center freq + BFO) / B = Change BFO frequency.

16 : Use this box to toggle **AGC on/off**. In AM mode when using Coh2, there are three options : Off = No AGC / AGC = AGC derived from peak power in the entire passband / COH = Independent AF-derived AGC for the left and right channel.

17 : Use this box to enter **time constant for AGC attack**. A long time constant will prevent the AGC from reducing the gain due to short bursts of interference. This parameter will affect the release time which can never be smaller than the attack time.

18 : Use this box to enter **time constant for AGC release**.

19 : Use this box to enter **hang time for the AGC**.

20 : **Squelch level**. Left click on this box to set the squelch level in dB. Set to zero to deactivate the squelch. The squelch is automatically inactivated if the baseband filter is wider than the baseband graph and the squelch is also deactivated when the number of points on the filter curve is too small. The squelch looks for peaks in the baseband spectrum. It is affected by the baseband bin resolution as well as by the number of averages. It is possible to make the squelch open on very weak CW signals by comparing the largest power bin to the noise floor, it can also reject carriers and CW signals by looking at tne Nth largest value. An SSB signal typically has many equidistant peaks due to its repetitive non sinusoidal waveform.

21 : **Squelch point**. Left click on this box to set the squelch point. The range is 0 to 99. The value 0 means that the bin with highest power will be used. 99 means that the median power will be used. Something like 10 will exclude one or two carriers, but open well on a voice signal. The squelch is automatically inactivated if the baseband filter is wider than the baseband graph and the squelch is also inactivated when the number of points on the filter curve is too small. The squelch looks for peaks in the baseband spectrum. It is affected by the baseband bin resolution as well as by the number of averages. It is possible to make the squelch open on very weak CW signals by comparing the largest power bin to the noise floor, it can also reject carriers and CW signals by looking at tne Nth largest value. An SSB signal typically has many equidistant peaks due to its repetitive non sinusoidal waveform.

22 : **Squelch time**. Left click on this box to set the squelch time in seconds. The range is 0 to 9. If you want the squelch to open before a transmission starts, set the parameter “Output delay margin” large. Maybe 2000 ms. Then set the squelch time to perhaps 4 seconds. Note that the FFT1/FFT2 storage time must be adequately long to hold the amount of data required. The squelch is automatically inactivated if the baseband filter is wider than the baseband graph and the squelch is also inactivated when the number of points on the filter curve is too small. The squelch looks for peaks in the baseband spectrum. It is affected by

the baseband bin resolution as well as by the number of averages. It is possible to make the squelch open on very weak CW signals by comparing the largest power bin to the noise floor, it can also reject carriers and CW signals by looking at the Nth largest value. An SSB signal typically has many equidistant peaks due to its repetitive non sinusoidal waveform.

23 : There are two different **implementations of the baseband filter and resampler**. 1) Back transformation from FFT3 where points are weighted according to the yellow curve in the baseband graph. This is the old implementation that was the only alternative up to Linrad-02.58. This method is by far most CPU efficient for wide filters with steep skirts. Signals that fall on the steep skirts get distorted in case the shape is determined essentially by the window function in use. Such distortion does not affect AM, CW or SSB reception, provided that the selected filter is much wider (20 times or more) than the frequency response of an individual FFT bin. This method is not suitable for wideband I/Q output through the network and it requires fairly large sizes for FFT3 and therefore it gives a substantial increase in the time delay from antenna to loudspeaker. 2) FIR filter with timf3 as input. This is the conventional SDR solution. It was introduced in Linrad-02.59. It is possible to select a baseband filter with only one bin and thereby produce a filter that is entirely determined by the window function in use for FFT3. Very small sizes for FFT3 can be used and therefore this method can provide a very short time delay from antenna to loudspeaker. This method can't be used for large FFT3 sizes at high baseband sampling speeds because it would overload the CPU.

24 : Press the left mouse button to move the **dB scale upwards**.

25 : Press the left mouse button to move the **dB scale downwards**.

26 : Press left mouse button to toggle the **baseband oscilloscope on/off**. The oscilloscope has a fixed position on screen, move other windows to the right hand side of the screen when using it. The oscilloscope displays the time function (timf3) used to produce the baseband spectrum (FFT3). The bandwidth of the signal displayed by the timf3 oscilloscope is set by the parameter "First mixer bandwidth reduction in powers of 2" among the mode dependant parameters. This display is intended for use while checking the timf3 noise blanker which is not yet implemented.

27 : **Number of notches** to apply in the baseband. This box is only available when the baseband filter is applied in the frequency domain (mixer mode = 1 upper right corner). Set to 0 to remove all notches. Set to a particular notch number to modify frequency or width.

Type in e.g. "1" instead of the "0" and you can then type in the position ("Pos") and width ("w") of the notch number 1. And so on if you need more notches (2, 3,...).

28 : Parameter "**Waterfall zero**". Press left mouse button and type in new value. This parameter selects the zero point (in dB) for the waterfall graph.

29 : Parameter "**Waterfall gain**". Press left mouse button and type in new value. This parameter selects the gain for the waterfall graph.

30 : Press the left mouse button to **half the third FFT size**. The current FFT3 size is indicated by it's power of two below this box. If you cannot reduce FFT3 size further, expand the frequency scale with the box in the left upper corner of this window or reduce window size. When the third FFT size is reduced, the frequency scale is contracted by a factor of two. The time delay caused by the third FFT is also halved.

31 : Press the left mouse button to **double the third FFT size**. The current FFT3 size is indicated by it's power of two below this box. When the third FFT size is doubled, the frequency scale is expanded by a factor of two. The number of data points describing the filter is also doubled which will allow twice as steep filter skirts. The time delay caused by the third FFT is also doubled, twice as many data points have to be collected before the transform can be computed. To expand the frequency scale without making the FFT size larger, use the buttons in the left side of this window.

32 : Mouse on the baseband spectrum (**BFO and filter control**). With a wheel mouse, fine tuning can also be made here. These controls are active only within the spectrum display area The upper red vertical line is the BFO frequency in the correct frequency scale. Grab the BFO by placing the mouse curser on it, then press the left mouse button. Then move it while keeping the button pressed. The two other red vertical lines are the BFO placed 10 respectively 100 times closer to the center frequency of the passband. The “zoomed in” BFO controls have to be used when a narrow passband is chosen causing the true frequency of the BFO to come outside the frequency range of the graph. The filter in use is shown in yellow. The filter shape is controlled by two yellow vertical lines. The upper yellow line sets the flat bandwidth while the lower yellow line sets the skirt steepness.

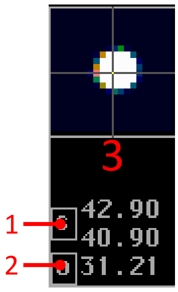
The baseband graph has two indicator numbers at the right hand side. These two numbers are not mouse control boxes. The upper is the FFT3 size in powers of two, the lower is the currently selected baseband processing mode which will change according to the settings in the control boxes. Use this number as your “Default output mode” in the parameter selection to have your favourite processing mode as the default mode.

The baseband waterfall has two numbers above each other on the right hand side. They form the two digit number that sets the step size when a wheel mouse or the left/right arrow keys are used for tuning. The step size is given in powers of two. Change the step size with the keyboard “+” or “-”. There are three tuning modes.

33 : Displayed when the notch is activated (27). **Frequency of notches** in the baseband. This box is only available when the baseband filter is applied in the frequency domain (mixer mode = 1 upper right corner). The number of notches must also be nonzero. Fraction of baseband filter width from -999 to 999.

34 : Displayed when the notch is activated (27). **Width of notches in the baseband**. This box is only available when the baseband filter is applied in the frequency domain (mixer mode = 1 upper right corner). The number of notches must also be nonzero. Fraction of baseband filter width from 0 to 99. Set to -1 to remove the selected notch.

3.5.2. Coherence window (par_xxx_cg)

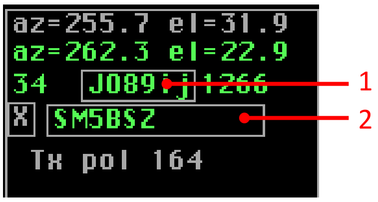


1 : Click this box to **show/hide the S-meter graph**.

2 : Click this box to **toggle the coherent oscilloscope on/off**. The coherent graph oscilloscope shows various signals in the baseband. These oscilloscope images are intended to assist in the development of automatic detect algorithms for CW and various digital modes.

3 : The **three S-meters in the coherent graph** show the strength of the signal that has passed the selected baseband filter. Set the filter to rectangular with 1kHz bandwidth to get the noise floor in 1000Hz. Then subtract 30 dB to get dB/Hz. The three meters show: Top = Peak S-meter / Middle = Current S-meter value / Bottom = True RMS for everything that passes the filter. The S-meter shows the level of a CW signal that has a speed that fits the selected passband. It is obtained from a bi-directional fast attack, slow release amplitude follower. The S-meter gives a level for CW signals that is independent of the mark to space ratio of the received signal. By pressing F2 you can get averaged true RMS readings above the normal three S-meter readings. Clear and restart with F2. The top line is the number of averaged values.

3.5.3. EME window (par_xxx_eg)



1 : Use this box to enter the **ww locator of another station** to get the optimum TX polarization and some more info about the DX location (in green, the azimuth & elevation of the moon at that location, as well as the azimuth, distance and spatial polarization offset from your own location).

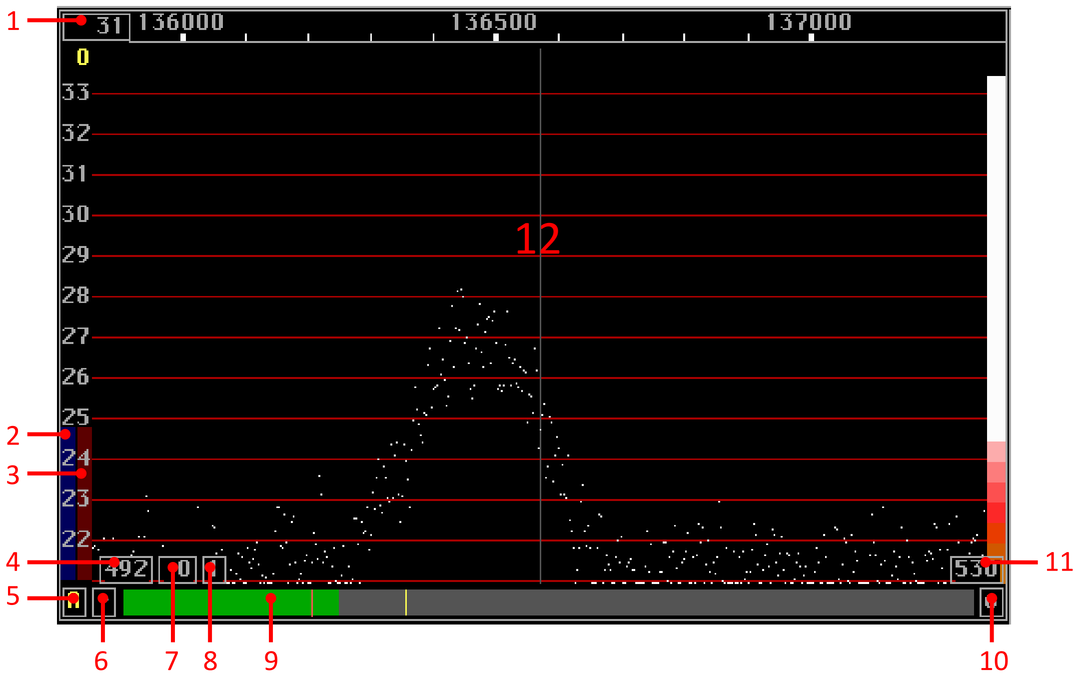
2 : Use this box to **enter a call sign or fragments thereof**. Linrad will search it's database and give the optimum TX polarization and some more info about the DX location in case the search gives a single station and the database contains the geographical location. Few parameters are stored in the file “own_info” which is located in the directory “C:\emedir” (see also the end of the section 2.2.).

3.5.4. Frequency window (par_xxx_fg)

1 : Left click this box, then enter a new **center frequency in MHz** for your receiver hardware and to display it on the frequency scales in Linrad. Right click this box to change the **step size** when the freq. is changed with the arrow boxes. Enter the desired frequency step in MHz (must be in the range 0.005 to 1.5). If a converter is used, enter the frequency of the output from the converter (= the frequency actually wanted as the center frequency of your SDR hardware.) The frequency scales will show frequency of the input to the converter by adding or subtracting the converter LO frequency and inverting the scale in case the LO is above the signal frequency. The default routine will fit the WSE converters, SDR-14, SDR-IQ, Perseus, Si570 (Softrock and others), Soft66, SDR-IP, xcalibur, hardwares with ExtIO libraries and perhaps more. Read the file z_USERS_HWARE to find out how this control can be configured to fit your own hardware in case you use something else.

2 : Click this box, then enter a new **gain value** for your hardware. The default routine will fit the WSE converters. Read the file z_USERS_HWARE to find out how this control can be configured to fit your own hardware in case you use something else.

3.5.5. High-resolution window (par_xxx_hg)



1 : Parameter "**fft2 avgnum**". Press left mouse button and type in a new value. This parameter controls the averaging of the high resolution spectrum, FFT2, which has red lines for the dB scale. Larger values give higher sensitivity but slower response.

2 : Selective **limiter level control for the first FFT**. Strong signals are prevented from reaching the blankers by the selective limiter. Frequencies that are not routed through the blankers are coloured red in the main spectrum (blue dB scale). Signals that are strong enough to be visible over interference pulses in the main spectrum are found very fast in the first FFT. The level is set by clicking the left mouse button in the bar control area. The first FFT averaging parameters affect the spectrum as you can see on the screen. Large averaging numbers gives a slower but more accurate determination of strong signals. Look at the spectrum, if parts of the noise floor becomes red, one of the limiter controls is too low or some averaging number is too small.

3 : Selective **limiter level control for the second FFT**. Strong signals are prevented from reaching the blankers by the selective limiter. Frequencies that are not routed through the blankers are coloured red in the main spectrum (blue dB scale). The second FFT is made from the signal after the noise blanker. This spectrum may have much better S/N due to the elimination of interference pulses. It is used to find strong signals that could not be found in the main spectrum because they are below the level of the interference pulses. The second FFT averaging parameter "fft2 avgnum" affects the spectrum as you can see on the screen in the high resolution graph (red dB scale). Large averaging numbers gives a slower but more accurate determination of strong signals. Look at the spectrum, if parts of the noise floor becomes red, one of the limiter controls is too low or some averaging number is too small.

4 : This parameter is 500 by default. Change it to **shift the high resolution spectrum upwards or downwards**. Note that the zero point for the colour scale in the main waterfall is the primary control for the vertical shift.

5 : Button to select mode for the **dumb blanker**. Toggle between modes by clicking the left mouse button. "-" = OFF / "A" = AUTO / "M" = MANUAL.

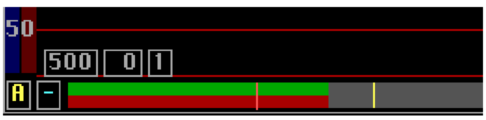
6 : Button to select mode for the **smart blanker**. Toggle between modes by clicking the left mouse button. "-" = OFF / "A" = AUTO / "M" = MANUAL.

Note that your system must be calibrated for the smart blanker to become available.

7 : **Attenuator use with MAP65**. It will attenuate the signal that Linrad sends to the network in timf2 format but it will have no effect on the processing within Linrad.

8 : Do not send any **strong signals to MAP65**. Signals on those frequencies that are red (defined with selective limiter level controls) in the main spectrum will not be sent to the network in the timf2 format.

9 : Mouse on blanker control bar. This bar shows the noise floor, the **noise power level** that remains after the blanker has removed pulses. The red vertical line is 20 dB above the quantisation noise of the internal data representation. Make sure your noise floor is not below this level. For setting signal levels, read the Linrad Home Page on the Internet. There are two noise blankers that are intended to be used together. Blue is the colour for the smart blanker while yellow is for the dumb one. When a blanker is enabled, the corresponding control level is indicated with a line on the blanker control bar. In MANUAL mode the control level is fixed and in AUTO mode it is held at some distance above the average noise floor. In both MANUAL and AUTO modes the control level can be changed by grapping the corresponding line with the mouse while the left button is kept pressed. Make sure to not set the control levels too low. The coloured numbers in the upper left and right corners give percentage of points cleared by the blankers. Keep these numbers small, the yellow below 20 and the blue below 4 unless you actually find a better readability of the desired signal with higher numbers. Hint : use the oscilloscope function to investigate what the blankers do to the noise that causes problems.



When 2 polarizations are selected, there are 2 noise power level bars.

10 : Button to **toggle timf2 oscilloscope**. The oscilloscope has a fixed place on the screen. To see it well, move other windows to the right side of the screen. The top track shows power. The next two tracks show I and Q of the "interesting" signal which is obtained from all frequencies that are coloured white in the main spectrum. The two lowest tracks show I and Q of the strong signals, those that are coloured red in the main spectrum. The strong signals are shown at a reduced amplitude, they are affected by an AGC function. The Y-scale of the timf2 oscilloscope can not be changed. If you have set the digital signal levels correctly, the scale will fit to the signals you see. Read about "Set digital signal levels correctly" on the Linrad Home Page on the Internet.

11 : This parameter is 500 by default. Change it to **increase or decrease the gain for the high resolution spectrum**. Note that the gain for the colour scale in the main waterfall is the primary gain control.

12 : Mouse on high resolution spectrum. Press the left button to select **the signal to process for loudspeaker output**. To eliminate a spur, place the mouse cursor on it and press E on the keyboard.

3.5.6. Meter window (par_xxx_mg)

1 : Press left button to **expand the vertical scale**.

2 : Press left button to **contract the vertical scale**.

3 : Press left button to **toggle scale between dB, dBm and S-units**.

4 : Press left mouse button to **decrease the averaging** in the meter graph. The averaging affects both the curves and the bar graph. The number of samples to form an RMS average over is a power of two This number is shown to the left. That number is also the number of samples out of which the largest value is picked to form the peak power graph.

5 : Press left mouse button to **increase the averaging** in the meter graph. The averaging affects both the curves and the bar graph. The number of samples to form an RMS average over is a power of two This number is shown to the left. That number is also the number of samples out of which the largest value is picked to form the peak power graph.

6 : Press left button to **move graph downwards**.

7 : Press left button to **move graph upwards**.

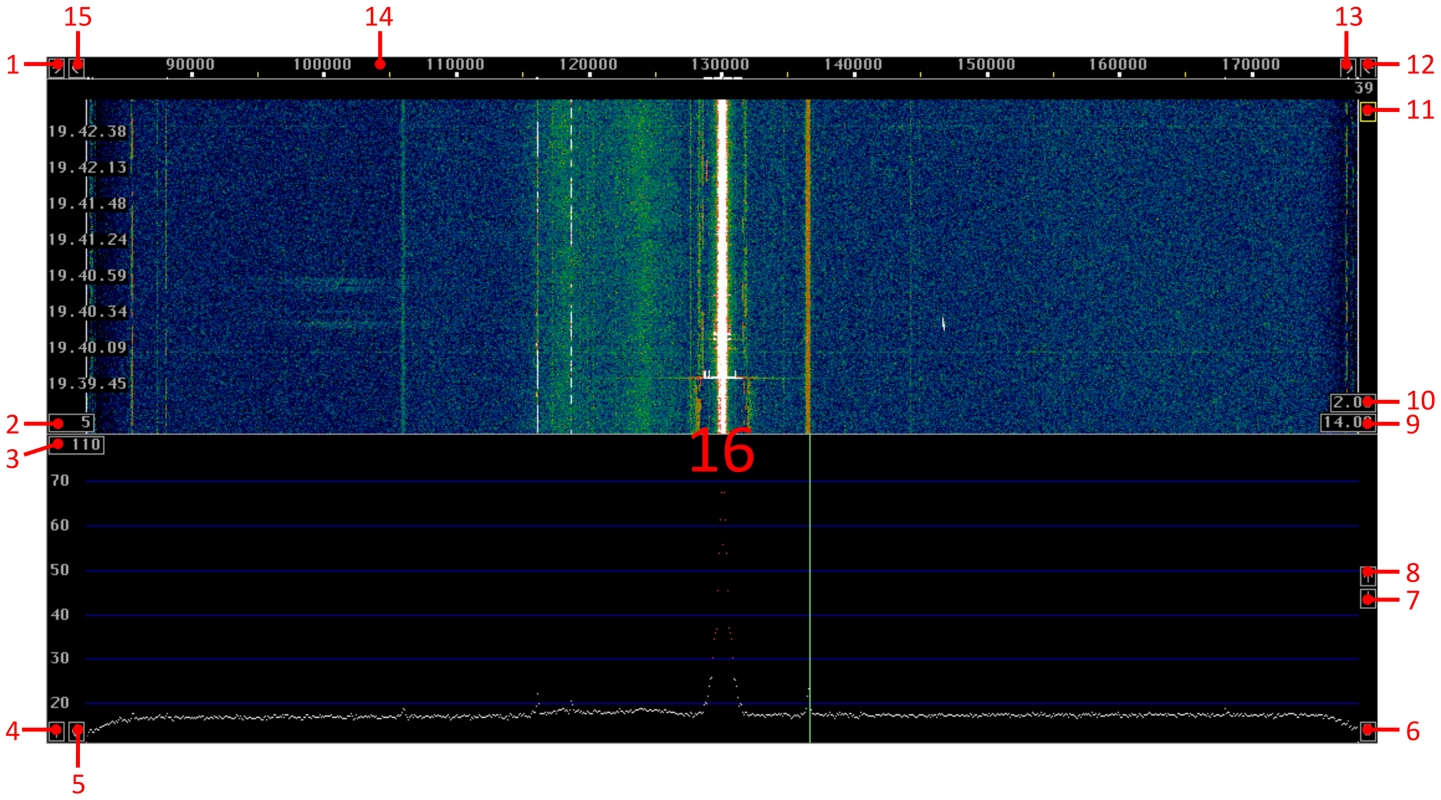
8 : Press left button to **toggle curves between** : P = **Peak** power / M = **RMS** power / 2 = Both **RMS and peak** power. Note that both the selected polarization and the orthogonal polarization is shown (in different colours) when two RF signals are fed into Linrad. The S-Meter bar shows only the selected polarization just like the power meter in the coherent graph.

9 : Press left button. Then type in a new **calibration constant in dB**. The dBm scale and the S-Units scale have separate calibration constants. The dB scale is the same as for the power meters in the coherent graph.

10 : The Linrad S-meter can show Peak power or RMS power in S-units, dB or dBm. Make the window small in the X-direction to see **signal strength as a bar graph** or make the window wide to see **level vs time curves**. Use the F1 help on the different boxes that appear when the graph is made wide to find out how to set the window to show what you want to see. With two RF channels, the S-meter shows the signal level from the

same two orthogonal combinations as shown in the baseband graph. Peak power is green and magenta corresponding to the two curves in the baseband graph just as white is baseband spectrum and S-meter peak power for a single channel. The RMS powers corresponding to each peak power are blue/green, red/magenta and yellow/white.

3.5.7. Waterfall window (par_xxx_wg)



1 : Press the left button to **contract the main spectrum and waterfall** by **moving in data points at the left hand** side. The highest frequency is left unchanged when the number of data points is large enough. Otherwise, the right hand side border line is moved inwards to make the window fit all the available data points.

2 : Parameter "**waterfall avgnum**". Press left mouse button and type in new value. This parameter controls the averaging for each line of the waterfall graph. Larger values give higher sensitivity but slower response.

3 : Parameter "**fft1 avgnum**". Press left mouse button and type in new value. This parameter controls the averaging of the main spectrum, FFT1, which has blue lines for the dB scale. Larger values give higher sensitivity but slower response. The parameter "fft1 avgnum" is always a multiple of the primary average number which can be changed by the box in the lower right corner of the main spectrum. When the first FFT bandwidth is high, the averaging process becomes a substantial part of the total computing and averaging in groups of up to 5 spectra saves time. The averaged FFT1 spectrum is used for the selective limiter. The sensitivity/speed compromise affects the noise blanker operation and the blue control bar at the left side of the high resolution graph must be set in accordance with the FFT1 avgnum value selected.

4 : Press the left button to **expand the main spectrum Y-scale**.

5 : Press the left button to **contract the main spectrum Y-scale**.

6 : The main spectrum is calculated as an average of the first FFT (FFT1). The **number of spectra to average** over is a multiple of the number in this box because averaging is done as averages of averages. Besides affecting CPU load at large numbers of averages, this parameter affects the noise blanker because the

blanker uses the average according to this number to locate strong signals. The blanker also uses the average of averages to locate strong signals closer to the noise floor. Press the mouse left button to change.

7 : Press the left button to **move the main spectrum downwards**.

8 : Press the left button to **move the main spectrum upwards**.

9 : Parameter "**Waterfall zero**". This parameter selects the zero point (in dB) for the waterfall graph. Press left mouse button and type in new value or right click to let Linrad auto-select the zero point. It will also affect the vertical position of the high resolution graph (red dB scale). The colour scale currently in use for the waterfall is displayed at the right hand side of the high resolution graph.

10 : Parameter "**Waterfall gain**". Press left mouse button and type in new value. This parameter selects the gain for the waterfall graph. It will also affect the dB scale of the high resolution graph (red dB scale). The colour scale currently in use for the waterfall is displayed at the right hand side of the high resolution graph.

11 : Click this button to enable or disable **automatic spur cancellation**.

12 : Press the left button to **contract the main spectrum and waterfall** by **moving in data points at the right hand** side. The lowest frequency is left unchanged when the number of data points is large enough. Otherwise the right hand side border line is moved inwards to make the window fit all the available data points.

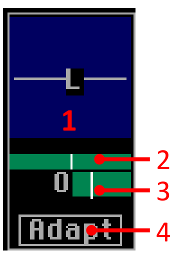
13 : Press the left button to **expand the main spectrum and waterfall** by **moving out points at the right hand** side. The lowest frequency is left unchanged when the number of data points is large enough. Otherwise the right hand side border line is moved inwards to make the window fit all the available data points.

14 : By clicking the frequency scale in the main spectrum/waterfall with the left mouse button you will enter the **zoom input mode**. The frequency scale and the arrow boxes at its sides will disappear and be replaced with two input fields where a lower frequency limit and an upper frequency limit can be typed in. Processing will not be changed until the center part (with the word "Apply") is clicked. When you click the frequency scale with the right mouse button, the frequency where you made the click will become the new center frequency to the extent that a shift is possible without getting outside the processed bandwidth.

15 : Press the left button to **expand the main spectrum and waterfall** by **moving out points at the left hand** side. The highest frequency is left unchanged.

16 : Mouse on main spectrum or waterfall graph. Press the left button to **select the frequency to process for audio output**. If mouse is on a frequency that is already selected the right button will deselect that frequency. To eliminate a spur, place the mouse cursor on it and press "E" on the keyboard. To clear spur elimination tables after a frequency change, press "C".

3.5.8. Polarization window (par_xxx_pg)



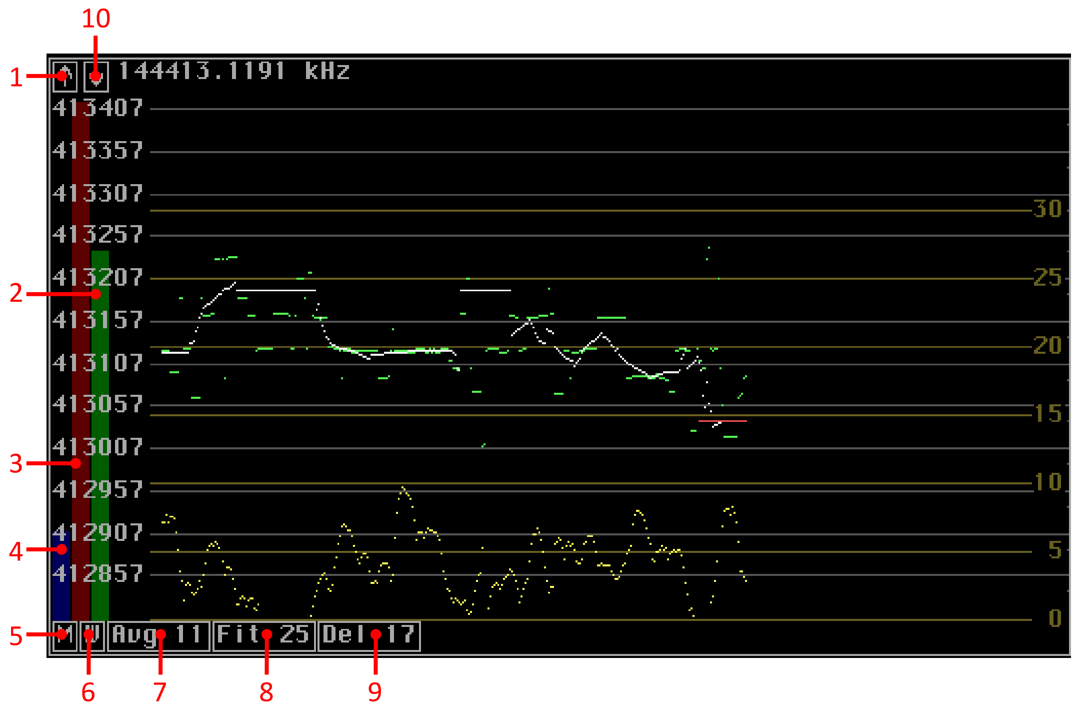
1 : Press left button to change **polarization angle**. This function is intended for use in "Fixed" mode.

2 : Press left button to change **ellipticity of polarization**. When the control line is at the extreme left or the extreme right within the box, the phase shift between the two incoming signals is 90 degrees. Then the resulting polarization can be set to circular, left or right hand rotation by changing the angle in the blue field above. When this control is at it's center position the polarization is linear. Together, angle and ellipticity can be set to match any polarization, linear, circular or elliptic.

3 : Press left button and move vertical line to **change time constant of polarization evaluation**. Only available in "Adapt" mode. For extremely weak signals with stable polarization such as 144MHz EME, use a long time constant. Signals that rapidly change polarization like 7MHz at night need to use a short time constant. Success of adaptive polarization is indicated by how well the signal is suppressed in the red spectra in the high resolution graph and in the baseband graph.

4 : Press left button **to toggle between "Fixed" and "Adapt"**. In "Fixed" mode the two incoming RF signals are combined to produce two output signals. The polarization angle (amplitude ratio) and the phase shift between the two RF signals can be set with the mouse in the two upper areas of this graph (use F1 for details). In "Adapt" mode, Linrad uses available information to adjust the polarization to fit the polarization of the incoming wave. In both cases the polarization is shown as a line, an ellipse or a circle. The displayed polarization is further processed and shown in green in the high resolution graph and in the baseband graph. The orthogonal polarization is displayed in red. Normally the orthogonal signal (red) is discarded, but by selecting the X+Y mode in the baseband one can get the two orthogonal polarizations in the two audio channels of a stereo output.

3.5.9. AFC window (par_wcw_ag or par_cw_ag)



1 : Click the left mouse button to make **AFC graph cover a smaller frequency range** and thereby decrease the AFC max lock range.

2 : **AFC search range limitation**. Use the left mouse button to change it. The max search range is set by the parameter "AFC lock range Hz" in the mode parameter setup. This bar is used to make the search range smaller in case the desired signal is close to some other (stronger) signal.

3 : **AFC lock range limitation**. Use the left mouse button to change it. Once the AFC has been successfully locked to a signal it will try to follow the signal even if it drops below the S/N value required for the initial locking. To prevent drifting away too far when locking to occasional peaks in pure noise in case the desired signal has paused for a while, the lock range is limited to half the frequency range selected for the AFC graph. This bar limits the lock range further.

4 : **Minimum S/N required in the search for a new signal**. Use the left mouse button to change it. Signals below this threshold do not lock the AFC to the frequency. In unlocked mode the AFC cursor in the high resolution graph is yellow. In manual mode it will stay on a fixed frequency until the mouse is clicked again to select a signal in the main spectrum or high resolution spectrum (blue or red dB scales). In automatic mode the AFC will continue to search for a signal above the required S/N value and lock to it as soon as one is found. In the search for a new signal, averages of the high resolution spectra (FFT2) are formed for the entire time during which spectra are available. The number of stored spectra is controlled by the parameter “Second FFT storage time (s)” in the mode setup. Average spectra are collected under the assumption that the frequency is drifting linearly with time. As a result, many average spectra are searched and the averaging made under the best assumption of frequency drift will give the strongest signal. The maximum frequency drift that will be searched for is set by the parameter “AFC max drift Hz/minute” in the mode parameters. The maximum frequency range that the AFC will search over is set by the parameter “AFC lock range Hz”. The search range can be reduced by the green bar at the right hand side of this control bar.

5 : Click the left button to **toggle AFC mode**. “-” = AFC disabled / “M” = MANUAL / “A” = AUTO (no yet implemented).

6 : Click the left button to **toggle time function window for spectrum averaging**. “-” = No window used. “W” = Use a sine(t) window. The window makes the AFC slower. It is intended to be used together with coherent averaging for extremely weak signals that can not be copied at all with “normal” methods.

7 : Number of spectra for the **AFC to average over while staying locked to a signal**. Click in the box with the left mouse button and type in a new value. The averaging is made without any concern of the frequency drift. This number has to be small enough for the peak caused by the signal to be on the same place over the entire period of time for which the averaging takes place. Check that the signal is not broadened much in the high resolution graph (red dB scale) when “fft2 avgnum” (see section 3.5.5.) is given the same value. For extremely weak but very stable signals, this number may be large which will allow an accurate frequency locking at the cost of a large time delay. The maximum number possible for this parameter is limited by the “First FFT storage time (s)” parameter in the mode parameter setup.

8 : **AFC straight line fit** (in number of spectra). Click in the box with the left mouse button and type in a new value. The primary AFC associates an average frequency to each spectrum. Each average is based on “avg” individual spectra which are summed with or without a sine window as set by one of the AFC control boxes. A straight line is fitted to these frequencies and the processing is made under the assumption that the straight line is correctly describing how the frequency changes linearly with time. A longer fit gives a better locking to stable signals and signals that drift linearly with time.

9 : **AFC time delay** (in number of spectra). Click in the box with the left mouse button and type in a new value. The AFC process analyzes spectra over a period of time. For weak signals (EME) that only occasionally are visible above the white noise floor, the AFC typically will fit the best frequency to 10 seconds of spectrum data. In such a case, best locking is obtained if the interpolated frequency is used, but then the delay caused by the AFC will be 5 seconds. By setting the AFC time delay to a smaller value than the maximum allowed (type in 999 for maximum), it is possible to make the delay smaller. The frequency used in processing will be less accurate, particularly for signals that drift randomly in frequency and are absent completely most of the time.

10 : Click the left mouse button to make **AFC graph cover a larger frequency range** and thereby increase the AFC max lock range.

3.5.10. CAT window (par_xxx_hwd_ug)

1 : Click on the arrows to select the computer serial **COM port** (RS-232 or USB to serial RS-232 converter) to which the transceiver is connected (CAT port of the transceiver).

2 : Mouse left click and type in the **raw offset frequency** between Linrad and the transceiver ; fine tune with the arrows if needed. This offset is used for example in case you use a 144/28 MHz transverter. If you have let's say "144.300" in the frequency window (see section 3.5.4.) of Linrad, you will need your transceiver to tune to "28.300" and the offset to be entered will be [Linrad frequency] + offset = [Transceiver frequency] ➔ offset = [Transceiver frequency] - [Linrad frequency] = 144.300 – 28.300 = -116.000 (don't type in the "." given here the offset is in kHz, while it is in MHz in the frequency window.

3 : Same as the raw offset frequency above but this one to compensate for **small frequency offsets** between the SDR hardware and the transceiver.

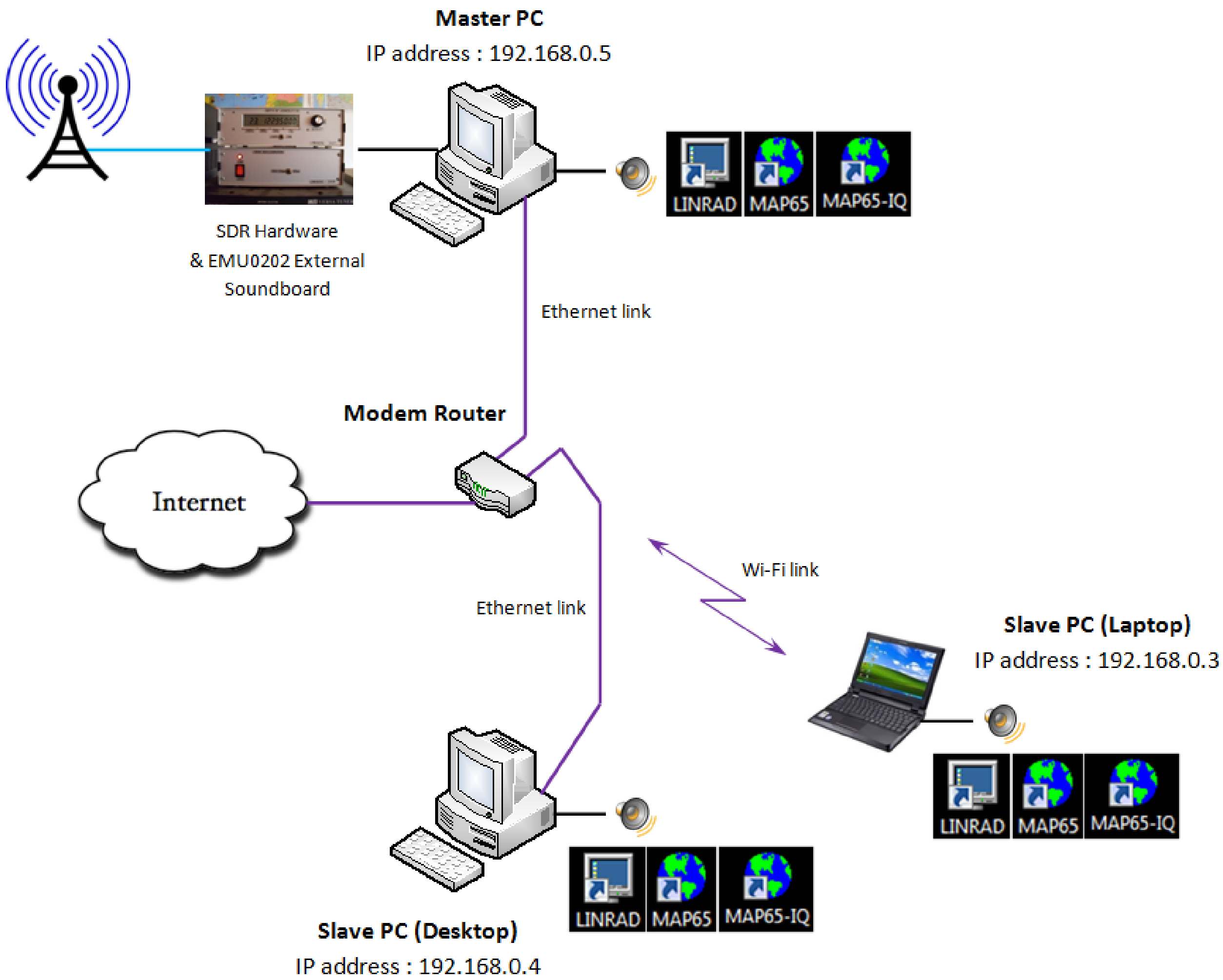
4 : Select your **transceiver type** with the arrows. The transceivers which are supported as is in the Linrad package are the following ones :

- **Kenwood**
  - TS-2000
  - TS-850
- **Icom**
  - IC-706
  - IC-275
  - "All Icom"
- **Yaesu**
  - FT-450
  - FT-736
  - FT-817
  - FT-897 (idem as FT-857)
  - FT-920
  - FT-950
  - FT-1000
  - FT-2000
- **Elecraft**
  - K3

4. Running Linrad in a network

It is possible to allow a Linrad instance running on a “master” PC to communicate with other Linrad instances on “slave” PC’s (running as remote consoles of the Linrad installed on the master computer). The same is applicable with MAP65, in the way Linrad on the master PC can directly communicate with MAP65 on slave PC’s (no need to have Linrad running on the slave PC’s if only for MAP65 purpose) . The master PC is the one connected to the radio hardware and the slave one is the remote PC. You have an architecture drawing below (IP addresses are just indicated as examples). The slave PC’s are only used for RX purpose.

4.1. Global architecture



4.2. Possible cases

The cases described here are "unicast" ones, i.e. one (master) PC communicating with one (slave) PC. It is also possible to do "mulicasting", i.e. one (master) PC communicating with several (slave) PC. I have not investigated this case ; it is then not described here.

	Master PC running with :	Communicates with	Slave PC running with :
Case 1	LINRAD	→	LINRAD
Case 2	LINRAD	→	MAP65
Case 3	LINRAD	→	MAP65-IQ

	Master PC running with :	Communicates with	Master PC running with :
Case 4	LINRAD	→	MAP65
Case 5	LINRAD	→	MAP65-IQ

4.2.1. Prerequisite (only for cases 1, 2 and 3)

You have to know the IP address of your computers. This is quite simple :

- You execute the same sequence than the one shown in section 2.3. at stage "Open the DOS window", until you get the DOS window.
- In this window, you type "ipconfig" + Enter, as shown below.

```
Microsoft Windows [Version 6.1.7601]
Copyright (c) 2009 Microsoft Corporation.  All rights reserved.

C:\Users\gho>ipconfig
```

```
   Media State . . . . . . . . . . . : Media disconnected
   Connection-specific DNS Suffix  . :

Wireless LAN adapter Wireless Network Connection 2:

   Media State . . . . . . . . . . . : Media disconnected
   Connection-specific DNS Suffix  . :

Wireless LAN adapter Wireless Network Connection:

   Connection-specific DNS Suffix  . :
   Link-local IPv6 Address . . . . . : fe80::f43f:4105:8196:8d78%11
   IPv4 Address. . . . . . . . . . . : 192.168.0.2
   Subnet Mask . . . . . . . . . . . : 255.255.255.0
   Default Gateway . . . . . . . . . : 192.168.0.1

Ethernet adapter Local Area Connection:

   Media State . . . . . . . . . . . : Media disconnected
   Connection-specific DNS Suffix  . : Enterprise.astrid

Tunnel adapter Teredo Tunneling Pseudo-Interface:

   Connection-specific DNS Suffix  . :
```

In the present example, the IP address of the computer is 192.168.0.2.

- Close the DOS window and perform the same operation on the other computers.

4.2.2. Case 1 : Linrad on master to Linrad on slave

4.2.2.1. The master PC

Linrad has been previously installed as described in section 2.2. or 2.3.

- In the main menu of Linrad, press "N" (Network set up). At this stage you will enter the IP address of the slave PC (the one that will receive the data from the master PC).
- Type "2" + Enter, then "-1" + Enter and finally the IP address of the slave computer + Enter, as shown below.

```
 CURRENT NETWORK SETTINGS ARE:

  1:      Base port = 50000
  2:     SEND address = 192.168.0.4
  3: RECEIVE address = 239.255.0.0
  4: Send raw data in 16 bit format OFF
  5: Send raw data in 18 bit format OFF
  6: Send raw data in 24 bit format OFF
  7: Send FFT1 transforms OFF
  8: Send timf2 (blanker output) OFF
  9: Send FFT2 transforms OFF
 10: Send baseband (resampled,16 bit) OFF
 11: Send baseband raw (24 bit)OFF
 12: RX input from network OFF
 F1: Help

On exit from this routine network transmit will be OFF.
The screen shows what will become enabled when send
is enabled (with T on the main menu)
(Do not forget to save with W on the main menu)

Enter a line number to change, 0 to exit => 2


          Default multicast SEND address range is    239.255.0.00 to 239.255.0.15

          Enter 0 to 15 to select an IP address within the default range
          Or enter -1 to specify a different IP address (p.e. 127.0.0.1): -1


          Specify different IP address: 192.168.0.4_
```

- You press on “4” + Enter.

```
 CURRENT NETWORK SETTINGS ARE:

  1:      Base port = 50000
  2:     SEND address = 192.168.0.4
  3: RECEIVE address = 239.255.0.0
  4: Send raw data in 16 bit format ON  (port=50000)
  5: Send raw data in 18 bit format OFF
  6: Send raw data in 24 bit format OFF
  7: Send FFT1 transforms OFF
  8: Send timf2 (blanker output) OFF
  9: Send FFT2 transforms OFF
 10: Send baseband (resampled,16 bit) OFF
 11: Send baseband raw (24 bit)OFF
 12: RX input from network OFF
 F1: Help

On exit from this routine network transmit will be OFF.
The screen shows what will become enabled when send
is enabled (with T on the main menu)
(Do not forget to save with W on the main menu)

Enter a line number to change, 0 to exit => _
```

- You press “0” + Enter to exit and come back to the main menu.
- In the main menu, press “T” (Toggle network output) and then “W” to save the parameters. Notice the “NETSEND” at the top left, indicating Linrad is sending data on the network.

```
NETSEND        Linrad-03.33   Soundcard
               expert mode

A=Weak signal CW             1=Process first file named in 'adfile'
B=Normal CW                  2=Process first file named in 'adwav'
C=Meteor scatter CW          3=Select file from 'adfile'
D=SSB                        4=Select file from 'adwav'
E=FM                         5=File converter .raw to .wav
F=AM                         T=Toggle network output
G=QRSS CW
H=TX TEST
I=SOUNDCARD TEST MODE
J=ANALOG HARDWARE TUNE
K=RADAR

M=Init moon tracking and EME database
N=Network set up
S=Global parms set up
U=A/D and D/A set up for RX
V=TX mode set up
W=Save current parameters in par_userint
F9=Emergency light
F1 or !=Show keyboard commands (HELP)
```

4.2.2.2. The slave PC

Linrad has been previously installed as described in section 2.2. or 2.3.

- In the main menu of Linrad, press “N” (Network set up).
- Type “12” + Enter, then “A”.

```
 CURRENT NETWORK SETTINGS ARE:

  1:      Base port = 50000
  2:     SEND address = 239.255.0.0
  3: RECEIVE address = 239.255.0.0
  4: Send raw data in 16 bit format OFF
  5: Send raw data in 18 bit format OFF
  6: Send raw data in 24 bit format OFF
  7: Send FFT1 transforms OFF
  8: Send timf2 (blanker output) OFF
  9: Send FFT2 transforms OFF
 10: Send baseband (resampled,16 bit) OFF
 11: Send baseband raw (24 bit)OFF
 12: RX input from network OFF
 F1: Help

On exit from this routine network transmit will be OFF.
The screen shows what will become enabled when send
is enabled (with T on the main menu)
(Do not forget to save with W on the main menu)

Enter a line number to change, 0 to exit => 12


          Set receive format.

          A=Raw 16 bit
          B=Raw 18 bit
          C=Raw 24 bit
          D=FFT1 transforms
          E=Baseband, resampled,16 bit
          F=Baseband, raw,24 bit
          G=timf2 (int16 or float32)
          Z=None
```

The RX input from network has now been activated.

```
 CURRENT NETWORK SETTINGS ARE:

  1:      Base port = 50000
  2:     SEND address = 239.255.0.0
  3: RECEIVE address = 239.255.0.0
  4: Send raw data in 16 bit format OFF
  5: Send raw data in 18 bit format OFF
  6: Send raw data in 24 bit format OFF
  7: Send FFT1 transforms OFF
  8: Send timf2 (blanker output) OFF
  9: Send FFT2 transforms OFF
 10: Send baseband (resampled,16 bit) OFF
 11: Send baseband raw (24 bit)OFF
 12: RX input from network ON (Raw data, 16 bit)
 F1: Help

On exit from this routine network transmit will be OFF.
The screen shows what will become enabled when send
is enabled (with T on the main menu)
(Do not forget to save with W on the main menu)

Enter a line number to change, 0 to exit => _
```

- Type “0” + Enter to come back to the main menu.
- In the main menu, type “U” to select the network as input.
- Type “A” (for the input), then “Y” (to select the network”). The window related to the network appears ; this has already been define above, so type “0” + Enter to exit.

```
CURRENT A/D and D/A SETUP FOR RX

Linrad RX input from: NETWORK
                      receive format     = RAW16

Linrad RX output to:  SOUNDCARD device   = Speakers (2- High Definition Au
                      device number      = 0
                      hostapi            = Native MME
                      min da sample rate = 8000
                      max da sample rate = 96000
                      min da bytes       = 1
                      max da bytes       = 2
                      min da channels    = 1
                      max da channels    = 2


DMA rate   min=10    max=30


A = Change the input settings and reset the output soundcard settings if a soundcard is selected as input.
B = Change the output soundcard settings.
C = Change min/max dma rate.
E = Enable/Disable frequency converter and set shift.
Z = Disable the output soundcard.
X = To main menu.
```

- Type “X” and press any key to come back to the main menu.
- Type “W” to save

4.2.3. Case 2 : Linrad on master to MAP65 on slave

4.2.3.1. The master PC

Linrad has been previously installed as described in section 2.2. or 2.3.

- In a similar way than in section 4.2.2.1., define the SEND (IP) address via menu “2”.
- Type “8” + Enter to send timf2 data to MAP65 via the network.

```
 CURRENT NETWORK SETTINGS ARE:

  1:      Base port = 50000
  2:    SEND address = 192.168.0.4
  3: RECEIVE address = 239.255.0.0
  4: Send raw data in 16 bit format OFF
  5: Send raw data in 18 bit format OFF
  6: Send raw data in 24 bit format OFF
  7: Send FFT1 transforms OFF
  8: Send timf2 (blanker output) ON  (port=50004)
  9: Send FFT2 transforms OFF
 10: Send baseband (resampled,16 bit) OFF
 11: Send baseband raw (24 bit)OFF
 12: RX input from network OFF
 F1: Help

On exit from this routine network transmit will be OFF.
The screen shows what will become enabled when send
is enabled (with T on the main menu)
(Do not forget to save with W on the main menu)

Enter a line number to change, 0 to exit => _
```

- Type “0” + Enter to return the main menu.
- Type “T” and “W” to save. Notice the “NETSEND” at the top left, indicating Linrad is sending data on the network.

4.2.3.2. The slave PC

MAP65 has been previously installed. In MAP65 (V2), in the menu “Setup” > “Options” > “I/O Devices” > “Network UDP Packets” must be checked.

4.2.4. Case 3 : Linrad on master to MAP65-IQ on slave

4.2.4.1. The master PC

Linrad has been previously installed as described in section 2.2. or 2.3.

- First proceed as in section 4.2.3.1.
- Then Type “1” + Enter
- And type “50020” + Enter because for MAP65-IQ, the base port 50020 is required, instead of 50000 for MAP65

```
  CURRENT NETWORK SETTINGS ARE:

  1:      Base port = 50020
  2:     SEND address = 192.168.0.4
  3: RECEIVE address = 239.255.0.0
  4: Send raw data in 16 bit format OFF
  5: Send raw data in 18 bit format OFF
  6: Send raw data in 24 bit format OFF
  7: Send FFT1 transforms OFF
  8: Send timf2 (blanker output) ON  (port=50024)
  9: Send FFT2 transforms OFF
 10: Send baseband (resampled,16 bit) OFF
 11: Send baseband raw (24 bit)OFF
 12: RX input from network OFF
 F1: Help

On exit from this routine network transmit will be OFF.
The screen shows what will become enabled when send
is enabled (with T on the main menu)
(Do not forget to save with W on the main menu)

Enter a line number to change, 0 to exit => _
```

- Type “0” + Enter to return the main menu.
- Type “T” and “W” to save. Notice the “NETSEND” at the top left, indicating Linrad is sending data on the network.

4.2.4.2. The slave PC

MAP65-IQ has been previously installed. In MAP65-IQ (V0.9), in the menu “Setup” > “Input data unicast” must be checked.

4.2.5. Case 4 : Linrad on master to MAP65 on master

Proceed the same way as in section 4.2.3. (Case 2), except the IP address to be entered (see section 4.2.2.1. towards which Case 2 refers) **must be 127.0.0.1** (and nothing else), given it is the IP address of an IP connection from a computer to the same (local) computer.

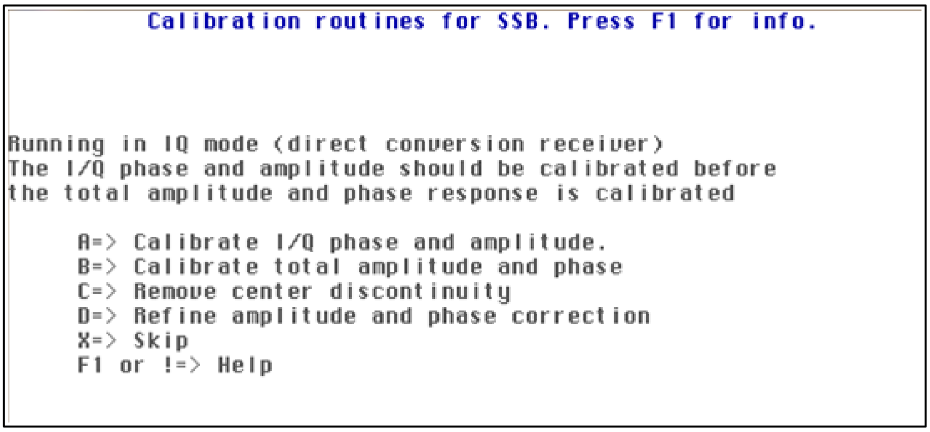
4.2.6. Case 5 : Linrad on master to MAP65-IQ on master

Proceed the same way as in section 4.2.4. (Case 3), except the IP address to be entered (see section 4.2.2.1. towards which Case 3 refers) **must be 127.0.0.1** (and nothing else), given it is the IP address of an IP connection from a computer to the same (local) computer.

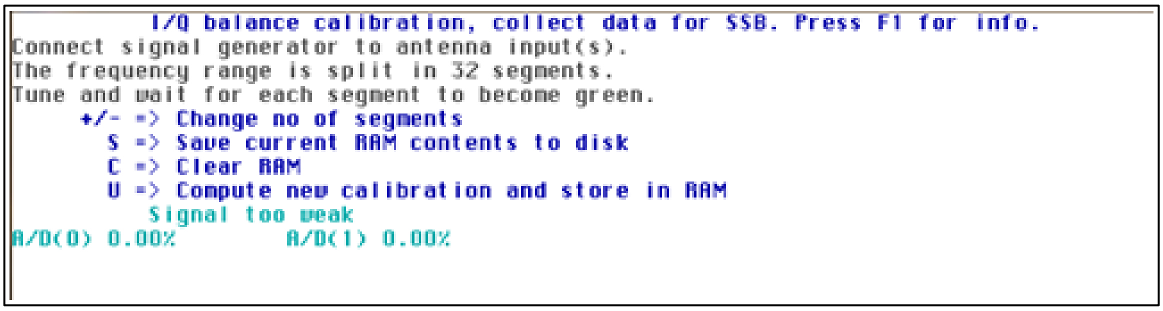
5. Calibrate Linrad

As in any SDR software, a calibration is needed in Linrad to attenuate as much as possible the image frequency, the one located "on the other side" of the center frequency, at the same frequency space than the useful signal. If the useful signal is in USB, at say 25 kHz on the right of the center frequency, the image signal will be in LSB, at 25 kHz on the left of the center frequency. If the I & Q signals would be perfectly in quadrature (90° of phase shift) and of exactly the same amplitude, the image frequency rejection would be infinite. In practice, it is not the case and the purpose of the calibration is to compensate for phase and amplitude unbalance to achieve to highest possible degree of image rejection. An advantage of Linrad over many other softwares is that the calibration is performed over the whole frequency span, through segmentation ; and not at a single frequency.

- When Linrad is running, type on "X", then "C" and you get a window like below.



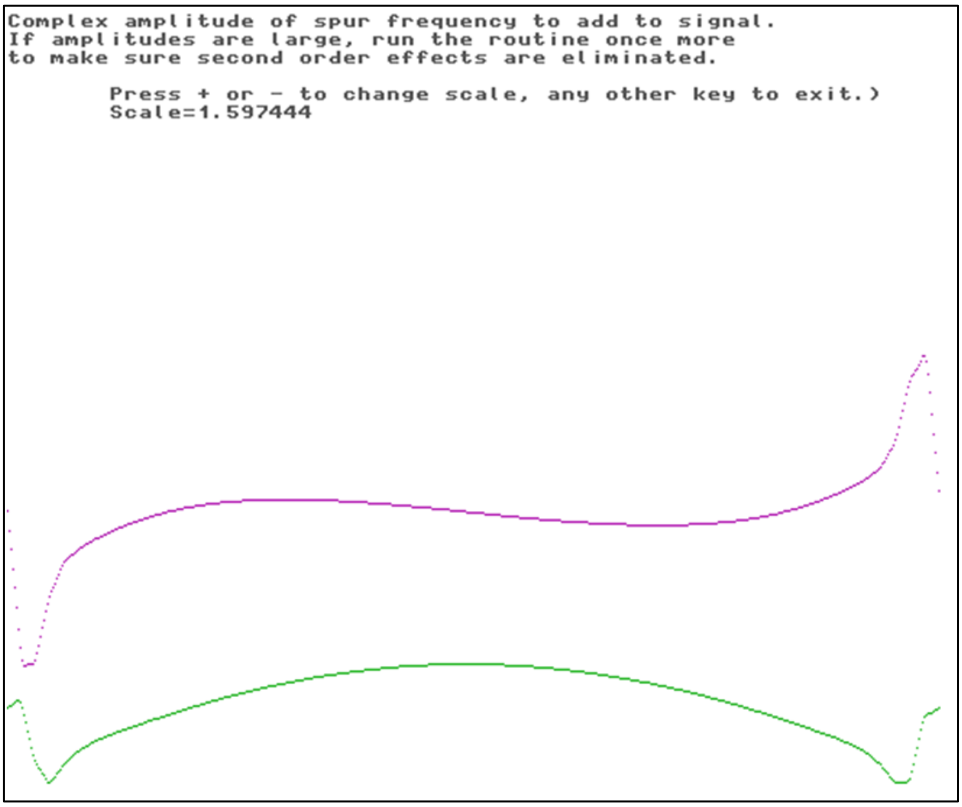
The "I/Q phase and amplitude" (A) calibration must be performed first and this requires a RF.



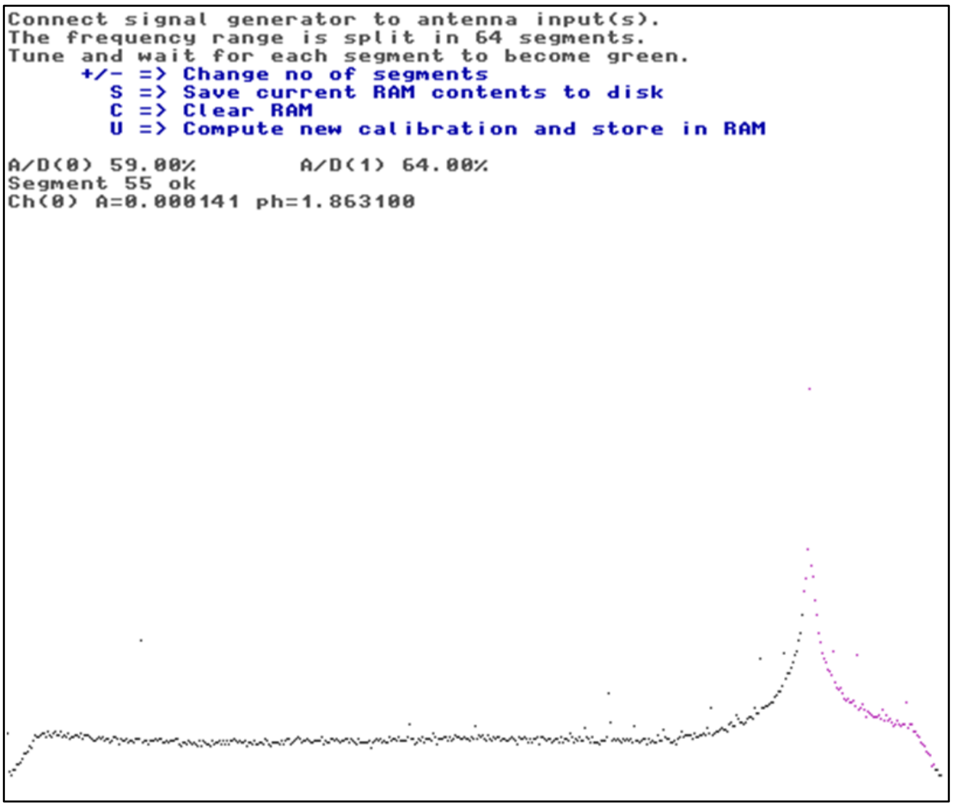
- Connect the RF generator to the SDR antenna input ; feed the same signal into both inputs in case you have enabled two RF channels.
- Adjust the level of the generator for about 50% level on the A/D converters. When everything is adjusted to your satisfaction, clear data that might have been collected while you were changing signal levels by pressing “C”. 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.

It is convenient to perform the calibration with a synthetized RF generator that can be set for a step length equal to the segment separation. With 64 segments and 96 kHz sampling, the step size is 1.5 kHz.

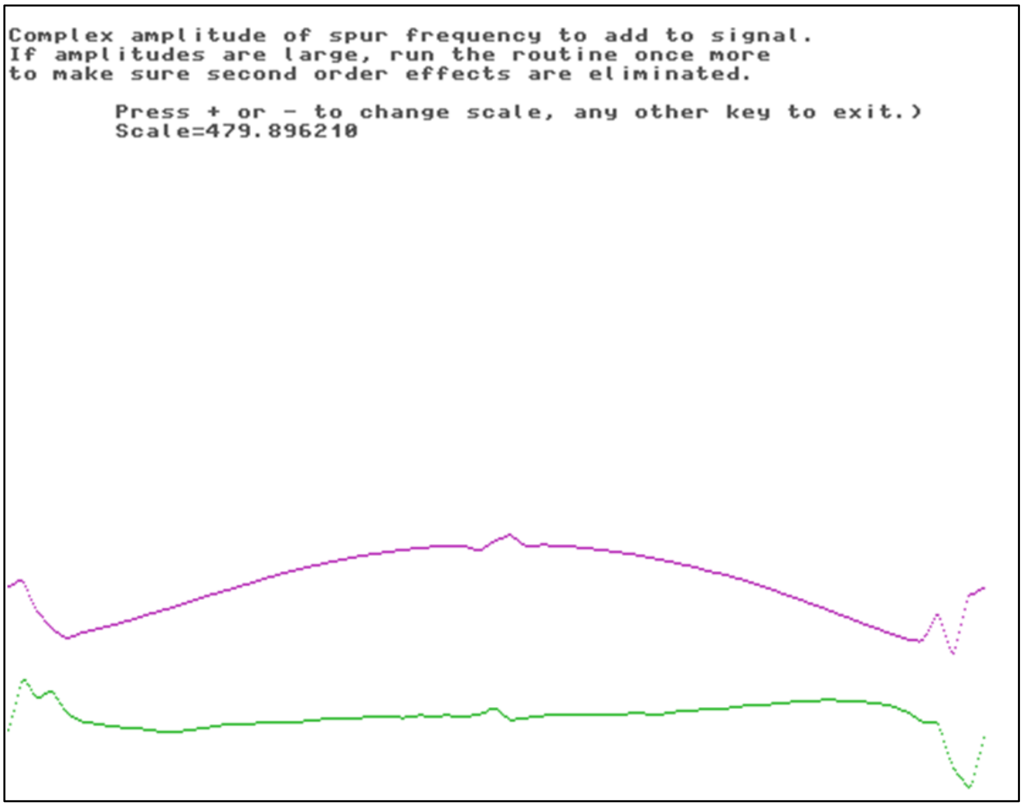
- The procedure is simple : step to another frequency, segment after segment. The points should turn from white to red. Then after a while they should turn green. Once the points on the current frequency are green, step to another white region. When all points are green, press “U” to compute the correction function. You should see a screen like the one hereafter.



- When you press Enter, the correction will be applied and you will return to the data input screen.



After a couple of iterations, the results no longer improve. The correction might look like in the figure below. The image suppression is typically 80 dB.



A correction file named “dsp_xxx_iqcorr” (xxx = mode) has now been placed in the directory where Linrad is installed.

The following calibration step (B in the calibration menu) requires a pulse generator (see http://www.sm5bsz.com/linuxdsp/flat/ampcal.htm). Once it is performed, a so-called “smart blanker” operation is possible.

I haven’t yet made the effort to perform the calibration beyond the “I/Q phase and amplitude” (A in the menu) step.

6. Parameter settings per usage purpose

In this section, one can find the parameter settings I use for different modes (SSB, CW, Weak CW for beacon monitoring, FSK441 or JT65), either for Linrad as a stand-alone receiver, as a remote receiver or together with WSJT9 and MAP65(-IQ). I don’t claim to be an expert ; I have spent many hour tweaking the parameters, so that now I have parameter sets meeting my requirements and environment, though there is still (and there will always be) room for improvement.

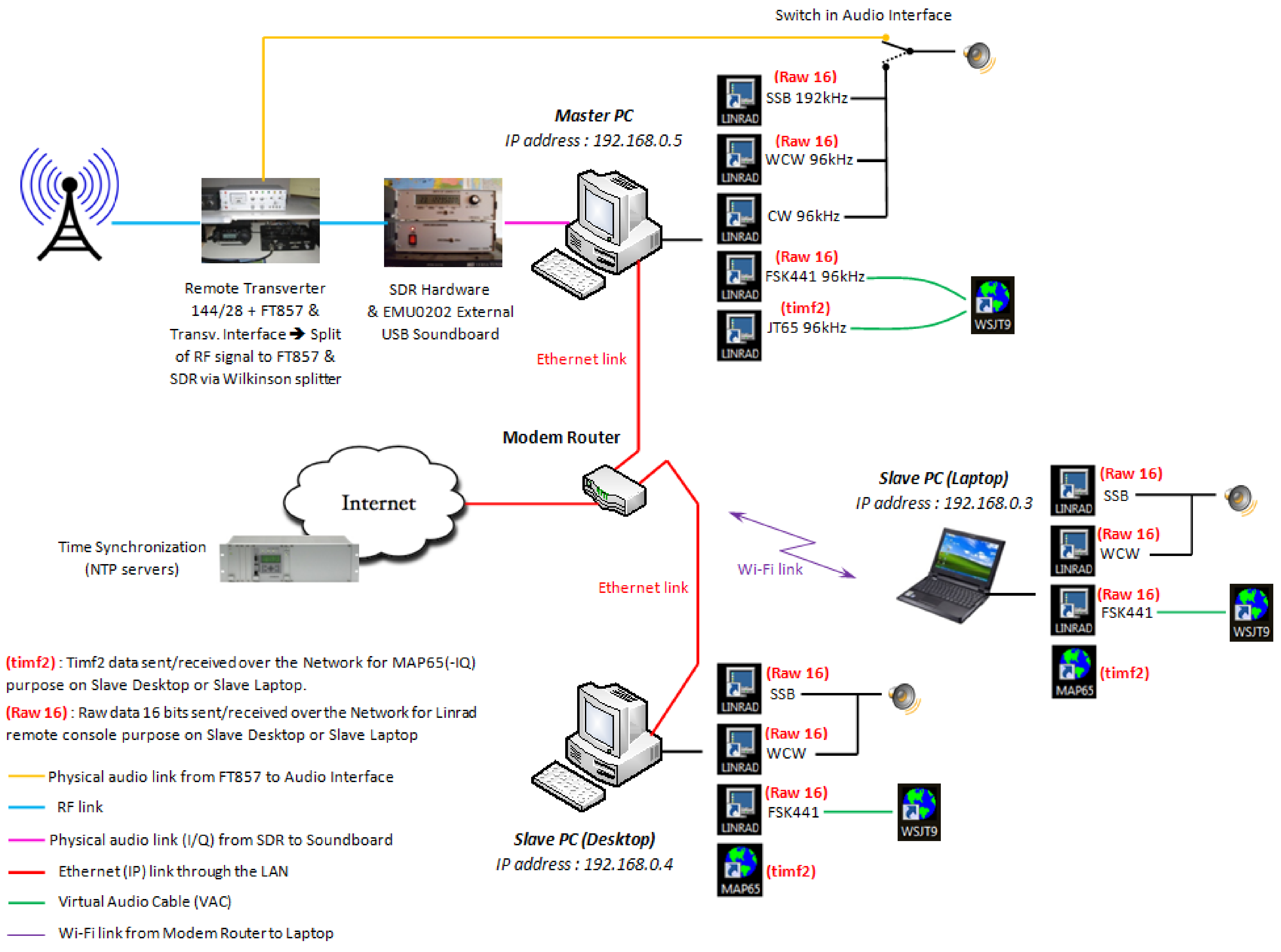
The parameter sets are provided in an apart document named “Linrad parameter sets Vx-y.xls”. The fact that document is separated from the present one facilitates the change tracking.

A more or less detailed description of the SDR hardware used (built around a Softrock) can be found on my website : http://www.on4khg.be/SDR.html.

6.1. Global architecture

The overall architecture schematic of my Linrad implementation is provided in the next page. It involves 3 computers, a master, a slave desktop and a slave laptop, in the same way as already shown in section 4 (Running Linrad in a network).

SDR System Architecture Drawing



6.2. Master PC

Brand :	GMT
Model :	
Processor :	Intel Pentium 4 CPU 3GHz
RAM :	2Go
Operating System :	Windows XP Family v2002 SP2
Screen resolution :	1280x1024
IP address (LAN) :	192.168.0.5

Name of the Linrad instance	SSB 192kHz	WCW 96kHz	CW 96kHz	FSK441 96kHz	JT65 96kHz
Soundboard sampling frequency	192kHz	96kHz			
Usage purpose	SSB	CW EME and Beacon monitoring	Normal CW	MS FSK441	EME JT65
Output Audio bandwidth (Hz) @ 0 dB	1900	60	150	1900	2400
Audio output target	Loudspeakers			VAC **(*)** to WSJT9	
Input	Soundboard E-MU0202				
Output	Soundboard Creative SB Live!			VAC **(*)**	
Processing delay (s)	0.7	6	1	0.7	0.7
Specificities	Sampling at 192 kHz to cover almost the full 144.200 to 144.400 MHz range	No AGC, slow waterfalls (high sensitivity)		Relatively fast waterfalls (for detection of the MS reflections). Baseband centered on the FSK441 tones	Sends data to MAP65(-IQ) on the slave PC's. Wide bandwidth for a flat noise floor in WSJT over 2,4 kHz
FFT1 size (pts) / Bandwidth (Hz)	4096 / 112.6	8192 / 23.4	4096 / 56.3	2048 / 112.6	2048 / 112.6
FFT2 size (pts) / Bandwidth (Hz)	65536 / 5.8	65536 / 2.9	8192 / 23.4	32768 / 5.8	32768 / 5.8
Data sent over the network	Raw 16 bits		None	Raw 16 bits	timf2

(*) : See http://software.muzychenko.net/eng/vac.htm#history. This is a shareware (cost ≈ 25€).

Actually, the master PC hosts twice the instances shown in the table ; one sends data over the network to the slave desktop and the other one sends data to the slave laptop. So, for example, on the master PC, there is a Linrad instance "SSB 192kHz" sending data to the IP 192.168.0.4 and another such named instance sending data to the IP 192.168.0.3.

6.3. Slave PC (Desktop)

Brand :	HP
Model :	P6033r
Processor :	AMD Athlon 64 X2 Dual Core 5000+ 2.6GHz
RAM :	4Go
Operating System :	Windows Vista Family Premium SP2 64 bits
Screen resolution :	1920x1080
IP address (LAN) :	192.168.0.4

Name of the Linrad instance	SSB	WCW	FSK441
Soundboard sampling frequency	Defined by corresponding Linrad instance on master PC		
Output Audio bandwidth (Hz) @ 0 dB	1900	60	1900
Audio output target	Loudspeakers		VAC (*) to WSJT9
Input	Network		
Output	On-board soundboard		VAC (*)
Processing delay (s)	0.7	6	0.7
FFT1 size (pts) / Bandwidth (Hz)	4096 / 112	8192 / 23.4	2048 / 112.6
FFT2 size (pts) / Bandwidth (Hz)	65536 / 5.8	65536 / 2.9	32768 / 5.8
Data received over the network	Raw 16 bits		

6.4. Slave PC (Laptop)

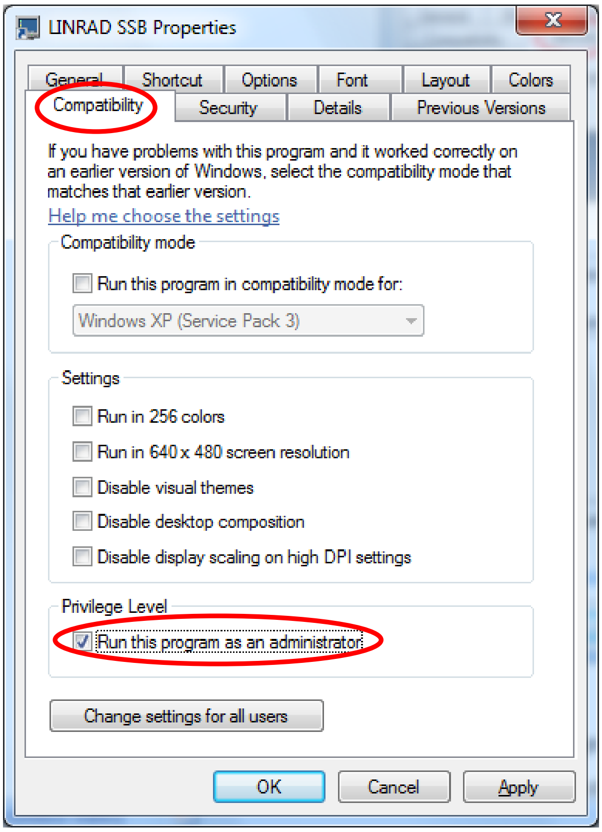
Brand :	Dell
Model :	Latitude E6400
Processor :	Intel Core 2 Duo CPU P8400 2.26GHz
RAM :	4Go
Operating System :	Windows 7 Professional SP1 32 bits
Screen resolution :	1280x800
IP address (LAN) :	192.168.0.3

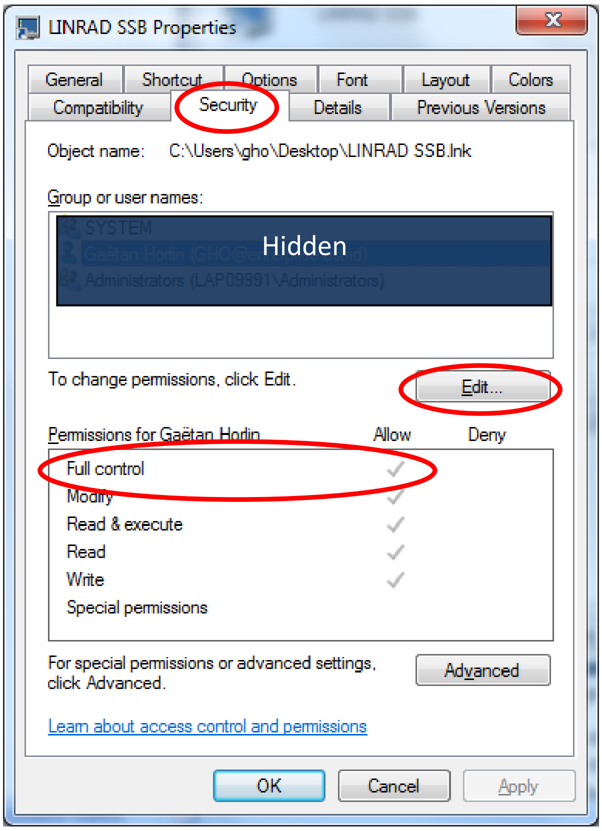
All the parameters are the same than on the Slave Desktop, except the XY coordinates (pixels) of the windows on the screen, since the screen resolution is not the same on both PC's.

7. Frequently Asked Questions (FAQ)

➢ Linrad in Windows Vista & Windows 7

Linrad should preferably be run as an “administrator” and “full control” of the directory where Linrad is installed is required. To run as an administrator, right click on the executable file “linrad.exe” (or whatever name you may have given) or its associated shortcut on the desktop, then left click on “Properties” and select the tab “Compatibility” in the new window. Check the box “Run this program as an administrator” (left picture):





Then select the tab “Security” and ensure the full control is allowed ; if not, activate it with the button “Edit”.

➢ How to restore a parameter set after a change in the GUI ?

When you are operating Linrad, there are situations where you need to change the width of the audio filter in the baseband window, change the zero of one of the waterfalls, and so on. When you will close Linrad, these changes will be reflected in the parameter files. If you don’t want these changes to be reflected in the parameter files, so that you always keep a “reference” parameter set, create a batch file (.bat).
Open the notepad and type in (for example) :

copy "par.*" "c:\Linrad Back-up Directory"*
Linrad_3-36_SSB_192k.exe

In the directory where the corresponding Linrad is installed, save as (for example) :

“Linrad 3-36 SSB 192k.bat”

Subsequently, always start Linrad by double clicking on this .bat file (you can create a desktop shortcut for it). This will first copy all the “par” files to the directory “Linrad Back-up Directory” and then it will start Linrad. This way, you can change whatever you want in Linrad while operating it.

After you have closed Linrad, just copy back manually (copy-paste) the “par” files from the directory “Linrad Back-up Directory” to the directory where Linrad is installed ; this will restore the previous “reference” parameter set.
Alternativeley, in the same way, you can imagine creating a batch file that automatically retrieves at every Linrad start-up the parameter files stored somewhere in a “reference” directory, by copying them towards the directory where the Linrad instance you want to start is installed.

In the batch file, if there are spaces in the names, these must be included between quotation marks (“…”) or underscores (_) must be used instead (and the associated file or directory names must be re-named accordingly).

How to start Linrad and another program at the same time ?

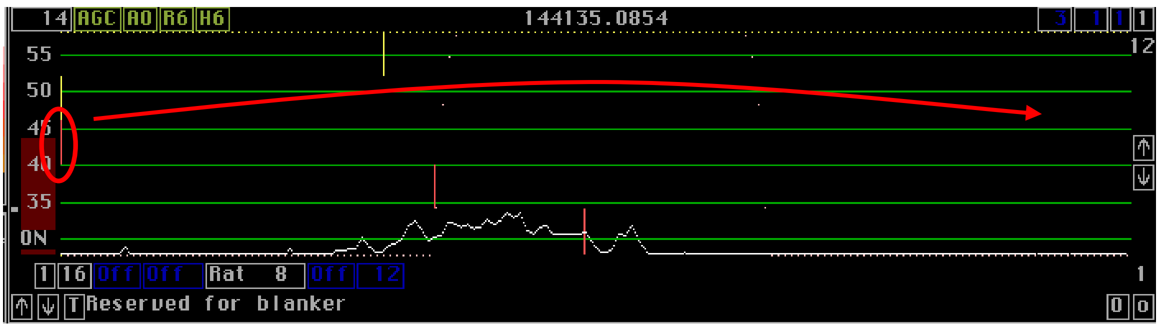
Imagine you want to start Linard together with MAP65. In the same way as in the preceding point (“How to restore a parameter set after a change in the GUI ?”), create a batch file that may look like (for example) :

```
copy "par*.*" "c:\Linrad Back-up Directory"
@echo off
start Linrad_3-36_JT65_96k.exe
start MAP65_from_Linrad.lnk
exit
```

When double clicking on this batch file (or it associated desktop shortcut icon), it will copy the “par” files to a “Back-up” directory (as in preceding point) and then start Linrad and MAP65. Actually, as one can see, MAP65 is started through a link (.lnk). This link is just a shortcut ; in the directory where MAP65 is installed, right click on “map65.exe”, then “create shortcut”. Move this shortcut to the directory where Linrad is installed (in the example, the directory where “Linrad_3-36_JT65_96k.exe” is installed).
Obviously, you can do the same thing with other programs, e.g. start Linrad, together with MAP65 and Live CQ (http://www.livecq.eu).

How to select USB or LSB ?

If you are in LSB and want to switch to USB, you can swap the I & Q channels at the soundboard input (but this has an impact on the direction of the frequency scale) or, in the baseband window, move one of the red vertical lines (BFO control, see 3.5.1. / 32) the other side to the one they stand.



If the picture above depicts Linrad is working in USB, move the red vertical line from the left to the right to select the other sideband (LSB), and vice versa from LSB to USB.

Linrad and MAP65 V2

1/ **Communication** Linrad > MAP65 ➔ see section 4.2.

2/ **Adjustement of levels** ➔ 4 Linrad parameters have an impact of the level towards MAP65 (beside any adjustement that may have been performed on the SDR hardware, like RX chain gain or soundboard level adjustment) :

- **First FFT amplitude** (in par_ssb file). Don't set it below "500" to avoid dynamic range issues.
- **First backward FFT att. N** (in par_ssb file). Mine is set to "6".
- **map65 gain** (in par_ssb_hg). Attenuator of timf2 for MAP65. Adjust for around 10 dB (see also param. "map65 strong" below) in MAP65. Mine is set to "15" (max. value).
- **map65 strong** (in par_ssb_hg). "0" = block strong signals to MAP65. The "strong" signals are the ones for which the points on the spectrum in the "Waterfall window" (see 3.5.7.) are coloured in red ; this is defined by the blue and red vertical bars (parameters "sellim fft1 S/N" and "sellim fft2 S/N" i n par_ssb_hg) on the left of the "High resolution window" (see 3.5.5.) When set to 0, the RX lev in MAP65 decreases by around 10 dB. "1" = allow strong signals to MAP65. Mine is set to "0".

If the level in MAP65 is too high, MAP65 is likely to freeze at some (random) point. My system is adjusted for a level around 10 dB in MAP65. With this level, no freezes have been experienced and decodes work down to -30 dB ("Normal deep search") ; this is as sensitive as WSJT9 stand-alone, if not even better.

3/ **Timing** ➔ since the "timf2" data stream is sent to MAP65, the processing delay introduced by Linrad is negligible. Indeed, "timf2" is the output of the first backward FFT, which is just the second stage within the whole Linard signal processing scheme.
However, for proper operation, MAP65 obviously needs a perfect synchronization of its own. With Windows XP, I use Dimension 4(http://www.thinkman.com/dimension4) and with Windows Vista or 7, I use Chronos Atomic Clock Synchronizer (http://www.chronosatomic.com) ; not a freeware but usable as such.

➢ Noise floor setting in Linrad

Linrad works preferably with a noise floor 20 dB above quantization noise. This achieved in most of the cases with a noise floor around 20 dB in the spectrum of the "Waterfall window" (see 3.5.7.).

8. Fast Fourier Transform “for Dummies”

This section will be completed as time being, in later releases of this document.

9. Revision history

Date	Revision	Version
25/01/2012	Creation of the document	0.1
29/01/2012	Review and addition of several sections	0.2
03/02/2012	Addition of section 4	0.3
05/02/2012	Several additions	0.4
25/02/2012	Update after feed-back of Leif, SM5BSZ. Global review and cosmetic modifications	0.5
08/03/2012	Addition of architecture drawings and section 6.	0.9
12/03/2012	Addition of section 7. Integration of feed-back from Leif. Global correction and review. First release.	1.0