

Linrad On-Screen Controls

K1JT

Main (Startup) Menu

A = Weak signal CW
B = Normal CW
C = Meteor scatter CW
D = SSB
E = FM
F = AM
G = QRSS CW
H = TX test
I = Soundcard test mode
J = Analog hardware tune

1 = Process first file named in 'adfile'
2 = Process first file named in 'adwav'
3 = Select file from 'adfile'
4 = Select file from 'adwav'

M = Init moon tracking and EME database
S = Global parameter setup
U = A/D and D/A setup for Rx
V = TX mode setup
W = Save current parameters in par_userint
Z = Hardware interface test
F1 = Show keyboard commands
ESC = Quit program

Keyboard commands (available in most modes)

A = Numerical display, amplitudes
C = Clear spur elimination tables (if AFC present)
E = Eliminate spur (if AFC present)
G = Save entire screen as .GIF file
S = Save input data as file (S again to stop)
T = Numerical display of processing delays
W = Write output data as wave file
X = Exit from current mode to menu
Z = Restart averaging for S-meter and numerical displays
F1 = Help (Place mouse on a screen object)
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 code intermediates)
F4 = Show RF envelope vs time as an oscilloscope function
ESC = Quit from program

Waterfall Graph

-> (top left): Left button to contract main spectrum and waterfall. HIGHEST frequency is left unchanged if entire spectrum will not fit in window.

<- (top left): Left button to expand main spectrum and waterfall. HIGHEST frequency is left unchanged if entire spectrum will not fit in window.

-> (top right): Left button to expand main spectrum and waterfall. LOWEST frequency is left unchanged if entire spectrum will not fit in window.

<- (top right): Left button to contract main spectrum and waterfall. LOWEST frequency is left unchanged if entire spectrum will not fit in window.

Box, bottom left: number of spectra to average for each horizontal line in waterfall.

Box, bottom right: Waterfall gain; also sets the red dB scale in high resolution graph. Color scale currently in use for waterfall is displayed at right of high-res graph.

Box, bottom right: Waterfall zero point; also sets vertical position of red dB scale in high resolution graph.

Mouse in waterfall graph or main spectrum: left button selects frequency for audio output. If mouse is on a frequency that is already selected, right button deselects that frequency. To eliminate a spur (if AFC is enabled), place mouse cursor on it and type E. To clear spur elimination tables after a frequency change, type C.

Main Spectrum (blue lines for dB scale)

Box, top left: "FFT1 avgnum". Press left mouse button and type in new value. This parameter controls the averaging of the main spectrum, FFT1. 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 saves time. The averages 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.

^ (lower left): expand dB scale for main spectrum

v (lower left): contract dB scale main spectrum

^ (middle right): move main spectrum up

v (middle right): move main spectrum down

Box, bottom right: The main spectrum is calculated as an average of FFT1. The number of spectra to average 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. Click to change.

Mouse in waterfall graph or main spectrum: left button selects frequency for audio output. If mouse is on a frequency that is already selected, right button deselects that frequency. To eliminate a spur, place mouse cursor on it and type E. To clear spur elimination tables after a frequency change, type C.

High Resolution Spectrum (red lines for dB scale)

Mouse click: select signal for audio output. To eliminate a spur, place mouse cursor on it and type E.

Box, top left: "FFT2 avgnum". Press left mouse button and type in new value. This parameter controls the averaging of the high resolution spectrum, FFT2. Larger values give higher sensitivity but slower response.

Vertical blue bar at left: Selective limiter 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 colored red in the main spectrum. Signals 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 give a slower but more accurate determination of strong signals.

Vertical red bar at left: Selective limiter 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 colored red in the main spectrum.

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 give a slower but more accurate determination of strong signals.

If parts of the noise floor become red, one of the limiter controls is too low or some averaging number is too small.

Box, bottom left (yellow): mode (Off/Auto/Manual) for dumb blanker
(Blue): mode (Off/Auto/Manual) for smart blanker
Smart blanker is available only if you have calibrated by using a pulse generator.

Blanker Control Bar (Bottom of High Resolution Graph)

Horizontal bars (green/red) at bottom: show the noise floor (the noise power that remains after the blanker has removed pulses), one bar for each polarization. The red vertical line is 20 dB above the quantization noise of the internal data representation. Make sure that your noise floor is not below this level.

The two noise blankers are intended to be used together. Yellow is used for the dumb blanker, blue for the smart blanker. When a blanker is enabled, its control level is indicated by a vertical yellow or blue line on the blanker control bar. In manual mode the control level is fixed; 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 dragging the yellow or blue line with the mouse.

Make sure not to set the control levels too low.

Yellow number, top left: percentage of points cleared by the dumb blanker. Keep this below 20 or so.

Blue number, top right: percentage of points cleared by the smart blanker. Keep this below 4 or so.

Hint: use the oscilloscope function to investigate what the blankers do to the noise that causes problems.

[o] (bottom right): Button to toggle on/off the timf2 oscilloscope. The oscilloscope has a fixed location on the screen. Top trace shows power summed over both receiver channels. Next four traces show I and Q (for both channels) of the "interesting" signal obtained from all frequencies colored white in the main spectrum. The lowest four traces show I and Q for the strong signals (those colored red in the main spectrum). The strong signals are shown at reduced amplitude, controlled by an AGC function. The Y-scale of the timf1 oscilloscope cannot be changed. If you have set the digital signal levels correctly, the scale will fit the signals you see.

Baseband Waterfall

Numbers at lower right: upper is FFT3 size (as a power of 2); lower is the currently selected baseband processing mode, which will change according to settings in control boxes. You can use this number to select your "default output mode" as a setup parameter.

-> (**Top left**): contract the frequency scale

<- (**Top left**): expand the frequency scale

-> (**Top right**): Click to double the size of FFT3. Current size is indicated (as a power of 2) at top right in the filter passband area. When FFT3 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 FFT3 is also doubled, since twice as many data points have to be collected before the transform can be taken.

<- (**Top right**): Click to halve the size of FFT3. Current size is indicated (as a power of 2) at top right in the filter passband area. If you cannot reduce the FFT3 size further, expand the frequency scale with the <- button at the top left, or reduce the window size. When FFT3 size is reduced, the frequency scale is contracted by a factor of two and the time delay caused by FFT3 is halved.

The spectrum produced by FFT3 is used for the secondary AFC.

If neither AFC nor coherent processing is enabled, there is little reason to have more than 4 points on the filter. For AFC only, 8 points is enough. When coherent processing is enabled 8 to 16 points will be enough for signals strong enough to decode by ear.

To expand or contract the frequency scale (within some limits) without changing the FFT3 size, use buttons at TOP left of this window.

Bottom left: "waterfall avgnum" for baseband graph

Bottom right: gain (upper) and zero point (lower) for baseband waterfall graph.

Baseband Spectrum (green dB scale)

Mouse click: controls BFO frequency and filter width and shape. Uppermost red vertical line indicates BFO frequency on the true frequency scale. The other red vertical lines are the BFO frequency placed 10 and 100 times closer to the center frequency of the passband. Use these "zoomed in" BFO controls when a narrow passband is selected and the true BFO frequency is outside the displayed frequency range.

Filter in use is shown in yellow. To change the flat bandwidth, drag the curve in the upper half of the graph. To change the filter steepness, drag it in the lower half.

Top left: "FFT3 avgnum". This parameter controls the averaging of the baseband spectrum, FFT3 (green lines for dB scale). Larger values give higher sensitivity but slower response.

The averaged FFT3 spectrum is used for the secondary AFC if AFC is enabled. The secondary AFC locates the maximum within 50% of the selected bandwidth and places the secondary AFC cursor at the peak position. The secondary AFC indicator is the green rectangle in the frequency scale of the baseband graph, visible only when AFC is enabled. The filter is automatically centered on the secondary AFC frequency although the yellow filter curve does not reflect this (to save some time for the graphics).

Note that the value selected for "FFT3 avgn" (should this be "FFT3 avgnum"?) directly affects the secondary AFC.

^ (right middle): move dB scale up
v (right middle): move dB scale down

Red bar at left: volume control bar. Click or drag the bar to move up or down. Note that its action may be unlike volume controls you are used to. Linrad (in weak CW mode) is designed to run without any AGC. Strong signals are amplitude limited. Bring this control to a maximum to get a saturated signal. Adjust your hardware (loudspeaker and/or soundcard output controls) for a loud but acceptable maximum sound level. This will be the sound level for saturating signals. When listening to weak signals, place the volume control bar so that the amplitude indicator (to the left of it) is between 25% and 50% of full scale. The amplitude indicator turns red when saturated and green when the signal is very close to zero.

Numbers at bottom left: Left is number of output channels (click to change, if applicable for selected mode). Right is number of output bits (leave it at 16).

Blue boxes near bottom:

1. Mode for audio compression or expansion: Off/Exp/Lim.

Off: in normal mode, output is identical to input up to the saturation level. An extremely fast AGC prevents the output from exceeding the saturation level, which leads to a limitation of amplitudes at the saturation level without the generation of overtones.

Exp: amplitude variations of the input signal are expanded by an exponential function. Intended for use only at bandwidths below 25 Hz, where the human ear can no longer discriminate frequency differences. The exponent used for expansion is a setup parameter.

Lim: output is equal to input up to the saturation level, and clipped at that level. Overtones will be generated when clipping occurs.

2. Mode for coherent processing:
 - Off: no coherent processing
 - Coh1: binaural CW
 - Coh2: I and Q go to left and right ear
 - Coh3: I to both ears

The coherent modes use a second filter that is "Rat" times narrower than the main filter shown in yellow. (More details in on-screen help.)

3. Rat is the ratio of bandwidths for the two filters

4. Toggle on/off delay between ears

5. Set delay value

6. Toggle "X+Y" mode on/off. X+Y mode routes the two receive signals to the two output channels. May be useful when polarization changes too quickly for automatic polarization adaptation to follow. Mode details in on-screen help.

^ (Bottom left): expand the dB scale

v (Bottom left): contract the dB scale

Number at bottom right: default output mode

-+o (Bottom right): Baseband oscilloscope gain and on/off controls

Coherence Graph

The three S-meters show the strength of the signal that has passed the selected baseband filter. Set the filter to rectangular with 1000 Hz bandwidth to get the noise floor in 1000 Hz. Then add 30 dB to get dB/Hz.

The three S-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 bidirectional 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.

Box at lower left [o]: Coherent graph oscilloscope shows various signals in the baseband. These images are intended to assist in the development of automatic detect algorithms for CW and digital modes. For details, see the code in coherent.c.