►Spectrum DSP M2 User Manual◄

C. Turner, KA7OEI ▲ R. Colclough GW8RDI
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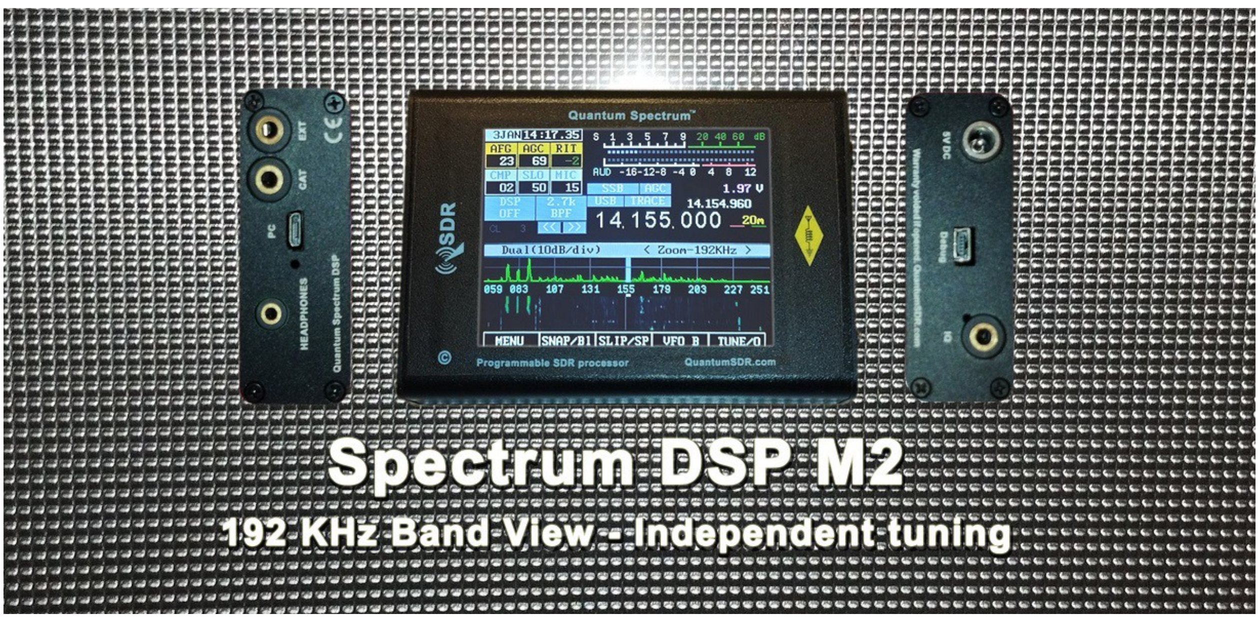
DRAFT FIRST RELEASE



Introduction:

The **Quantum SDR** project was formed to develop an affordable wide-band spectrum scope with waterfall that can be easily connected to any radio receiver or transceiver, incorporating the latest SDR technologies, including machine learning and artificial intelligence, dramatically improving access to the Spectrum for any radio system.

The team's aim is to continue enhancing the Spectrum software, and to develop a variety of new models, following the feedback we receive from users. You can contact us at info@QuantumSDR.com or through our Facebook group, QuantumSDR.

Thank you for your support.

The Spectrum DSP M2 incorporates the GUI (graphics user interface) originally developed by the Universal HAM SDR (Software Defined Radio) project and extended by Quantum SDR. It is reproduced here with their permission and our thanks.

Configuration: Touch interface and Transceiver/Receiver association

The Spectrum DSP arrives configured for the transceiver requested during ordering. There are two items that you may want to configure, one is the touch interface so that so that clicks and swipes align with the on-screen areas and buttons.

To realign the touch screen interface, reboot the Spectrum, and immediately that a small blue square shows, press the stylus on the screen and keep pressed. The TOUCH LINE TO RESET SETTINGS message should be displayed, move the stylus through that line. The touch calibration procedure should then start, else reboot and try again. Follow the on-screen instructions, finalizing by drawing through the green square, or if you want to repeat the calibration, through the red square.
If you press too soon after booting, the unit will perform a FACTORY RESET, following which you can reboot and calibrate the touch screen if needed.

Once set-up, click MENU and make a long-press on EXIT/* to save the new settings.

The other setting is to associate the transceiver or radio with which the Spectrum must connect via the CAT connection. This can be set by clicking the MENU button, then when the menu is displayed, make a short down-swipe in the middle-right of the LCD until Configuration Menu is reached, then another down-swipe on the right-side of the screen to open that menu. You can then select the associated transceiver and Spectrum mode, such as BiSync, slave or master.

NOTE YouTube instruction video

The LCD requires a light yet firm touch. Long-presses are activate when the stylus stays in contact over 0.6 seconds.
There is a video in our YouTube page where you can see how this is performed:
https://youtu.be/A_YlSJJQLGM?si=aTvMzkuZGbk8A-Cz

What radios can the Spectrum be used with?

The Spectrum DSP can connect to any radio system that has either an IQ port, IQ point on it's PCB (such as the Electaft KX2, uSDX (tr)usdx and newer QRP Labs radios), or a connection to the IF stage output (model M3 or higher).

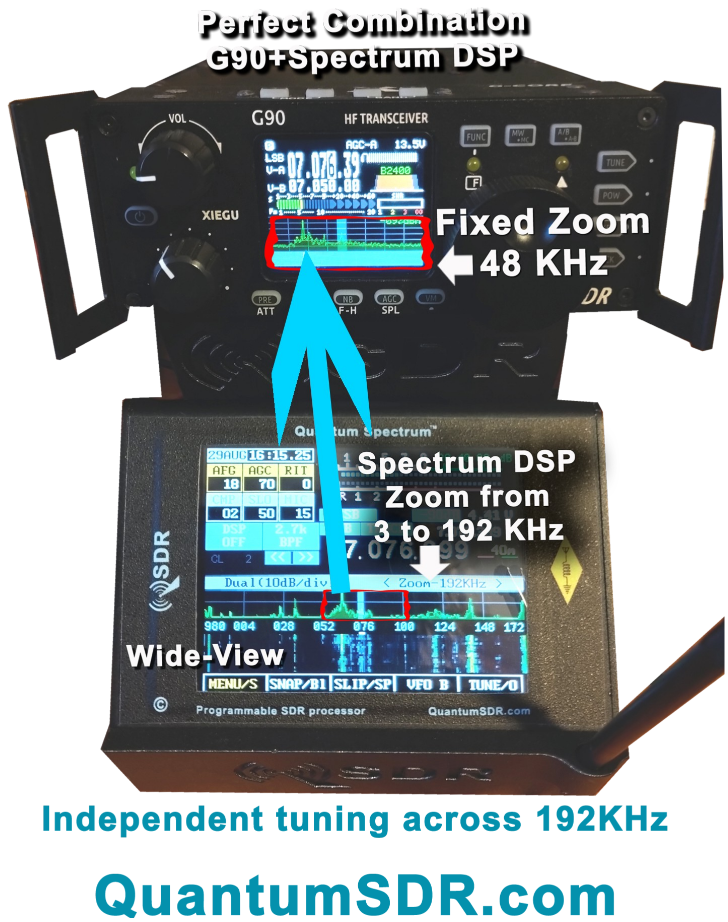
If you need support connecting your radio, please contact us.

Typical installations

Concept – Xiegu G90 to wide DSP

The Xiegu G90 is an excellent and feature-packed HF transceiver that's key limitation is it's small display. By the addition of the Spectrum DSP that small display is converted into the Zoom window, showing a close up of the stations nearest to tune frequency. Perfectly synchronized via the CAT connection, the Spectrum DSP becomes part of the station to provide not only a wide-band spectrum window that's zoomable from 3 to 192 Khz, but also an independent 192 Khz tuning window, where with the drag of the stylus, you can instantly listen to stations without having to spin the dial. When you find the station you want, one click is all it takes to synchronize the G90. DSP noise & notch filtering are also provided.

This combination results in a better option that upgrading to many of the newer DSP transceivers with larger screens that often lack items such as the ATU and full 20W output.



Concept – Elecraft KX3 and KX2 to wide DSP

The classic Elecraft KX2 and KX3, with it's beautiful smooth turning dial, synchronizes perfectly to the Spectrum DSP, providing an independent 192 Khz of instant tuning across the band with the stylus or a light touch on the screen. When you find the station that you want, one click synchronizes the Elecraft ready for operation. The Spectrum allows you to view the whole band, zooming from 3 right out to 192 KHz, and providing bidirectional tuning synchronization for the fastest access to stations across the bands.

Concept – Classic to wide DSP

The Spectrum DSP can also connect to the IF signal of any classic radio, allowing those beautiful sets to be used with the very latest features like any modern transceiver.

The connection to the IF of most receivers requires a simple miniature coax cable connection direct to the IQ input on the Spectrum. Connection notes are available for several models of radios.

If you require any help connecting your radio, please get in contact.

Graphical user-interface (GUI):

Note: All references to TX operation are included for the future inclusion of a TXRX module which converts the Spectrum into a DSP transceiver.

The touch screen is easily controlled once the basic technique is mastered.

Clicks with the stylus should be made in a deliberate manner, approximately half a second long will activate the menu and control buttons, any longer and they will be detected as a long-press, which activates secondary functions which most buttons and touch areas posses. You can practice this by clicking on the MENU button. A short but deliberate press will display the menu list, a long-press will mute the audio, making another long-press will reactivate the audio. If the volume is too low, click DSP/OFF and stroke up on the left-side.

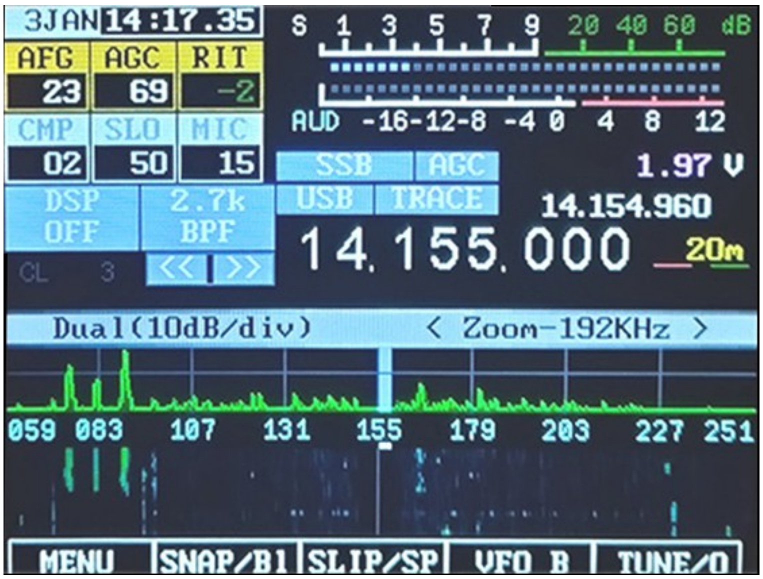


Figure 2: Main touch display of the Spectrum DSP

A video on YouTube is available showing how to use the touch interface, available here, https://youtu.be/A_YlSJJQLGM?si=0QFNF63dCD417fZ8

GUI controls are software defined, but for the purpose of simplicity they are typically defined as follows:

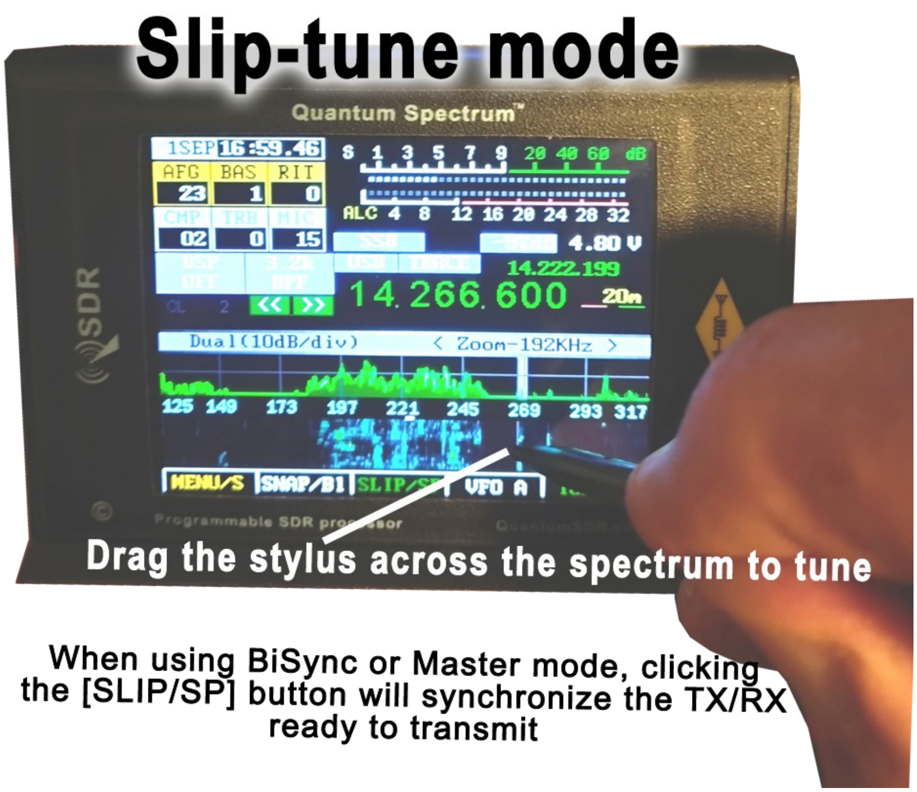
Important: SDR Offset

For best receive quality the frequency offset mode should be activated with a long-press on the TUNE/O button. When this button is green, the offset is activate.

- **Power on/off** – Tapping the LCD screen several times wakes the unit from power-save mode. To power-off, press Menu then click PWR and immediately Confirm.
- **Band change** – To change band click the highest part of the frequency display, which will then display the band keypad. You can also change band by clicking the red or green lines to the right-side of the frequency display.
- **Display Mode** – The display modes are toggled by clicking on the top of the Spectrum display area. If they gray header bar is shown, click on the left of the bar to toggle through the modes.

- **Tuning** – The Spectrum must be in Master, BiSync or Slip-Tune modes to enable tuning, else tuning is controlled from the attached transceiver when connect via CAT. Click MENU then select Configuration Menu to change mode.

  Tuning can be accomplished by clicking anywhere on the Spectrum band. To fine tune touch the centre of the spectrum display keeping the stylus down and drag left or right. Enabling SNAP/BI with a single click will increase the speed of frequency change through dragging. Fine tuning can be made with the two buttons **<< >>** a single press shifts the frequency 1KHz, a continuous press will continually bump the frequency by 100Hz. You can also click the right-most digits of the main frequency display to alternate between a 0 hertz offset, and 500 hertz, common channel steppings.
- **Slip-tune** – Click the SLIP/SP button once to enter and leave slip-tune mode. When activated the text will show green. When activated you can independently tuning by dragging the blue tune bar across the spectrum. The transceiver does not change frequency and the audio from the Spectrum will be heard and the frequency shown. Click SLIP/SP again will then synchronize the transceiver to the Spectrum frequency. A video on YouTube here explains the procedure: https://youtu.be/8nApkXsXI7A?si=YmrLRjyOvIi7Z6jf



A **video** on **YouTube** shows how to use the Slip-tune mode here.
Slip-tune allows **complete tuning freedom** over 192 Khz without re-tuning the TX/RX

- **Dial-tune** – Click the TUNE/O button bottom-right to display the virtual dial. Tune by drawing a circle with the stylus CW or CCW. To tune faster, make a long-press on the buttom button marked VFO A (or B), when this is shown red fast tuning mode is activated. The tuning is very accurate, and makes it easy to fine tune.

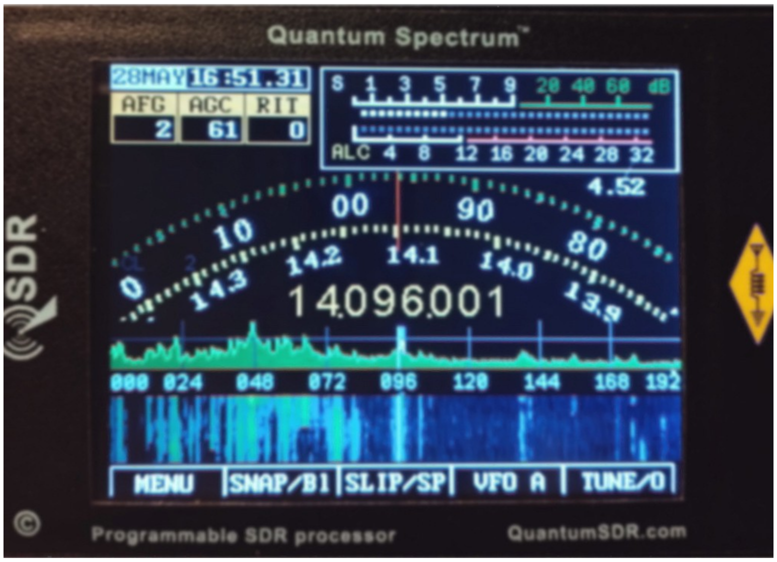


Figure 3: Virtual dial: The Spectrum after clicking the TUNE/O button.

- **Volume** – Click the DSP button centre-left then stroke up and down on the left-side of the LCD screen.
- **AFG** – Click the DSP button centre-left then stroke up and down in the middle of the display to adjust. Click the AGC box to change to BAS, TRB (treble) or AGC speed (SLO etc), then stroke up and down in the middle of the LCD to adjust.
- **RIT** – Click the DSP button centre-left then stroke up and down on the right-side of the screen to adjust RIT (Receiver Incremental Tuning), note that its function may be changed by clicking the RIT box to adjust the sending speed (in Words Per Minute) in the CW mode, or to adjust the Microphone or Line-In gain in voice mode, etc.
- **Mode** – Click the mode box in the center of the LCD to select the operating mode of the transceiver (CW, USB, LSB, etc.) Clicking this button cycles through the available modes. Click the button above to select digital modes, such as BPSDK and RTTY. When "**LSB/USB Auto Select**" is enabled in the Standard Menu (click MENU to access), the sideband that is not appropriate for the frequency of operation (e.g. USB will not be selected below 10 MHz). When menu item "**LSB/USB Auto Select**" is enabled, in order to change to AM you must select a mode ***other*** than LSB (or USB) – such as CW – and then press-and-hold the Mode box **-** AM will then be selected.
- **DSP** – This button is used to control the DSP audio filter mode. When the DSP keypa is displayed, pressing-and-holding will turn DSP on/off while preserving the current settings. A long-click on the DSP mode will also disable it until long-pressed again.
- **Filter button** – (Middle-left) This button is used to select the audio passband filter of the receiver. The button cycles through the filter that have been pre-selected in the Filter Selection menu. The button is disabled when FM mode is selected, but pressing-and-holding it while transmitting on FM will cause a tone burst to be generated, if this feature is enabled. Pressing-and-holding this button will force the selection of bandwidth that are otherwise disabled.

The row of virtual buttons at the bottom of the LCD screen are "soft" buttons, the functions of which change depending on mode, as indicated on the LCD, and will be discussed in more detail later in this document.

Main LCD Touch display

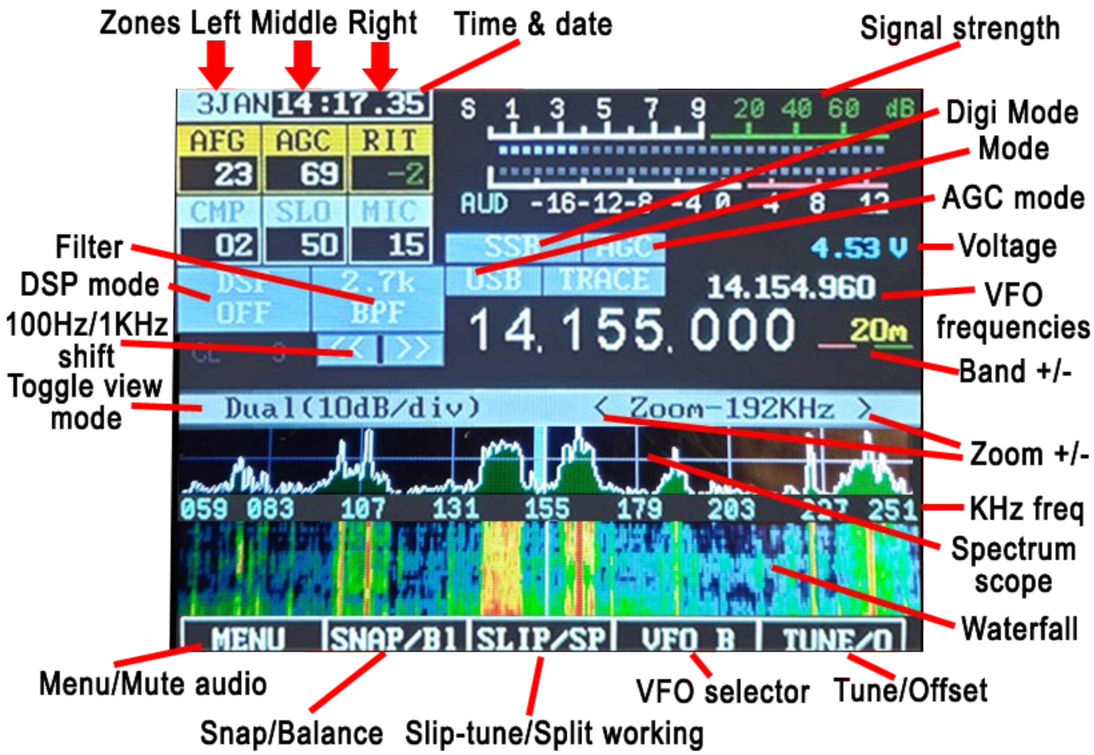


Figure 4: The main screen (annotated) with the SPLIT function disactivated.

When "Frequency Translate" mode is on, the center frequency indicator will be shifted right of center by 12 kHz to reduce the SDR noise level near DC zero.

On the main display, just above the Spectrum Scope, there are a number of indicators:

- **Main Frequency Display:** This may be displayed either as a single frequency *(transmit/receive as in **Figure 4**) or as a "Split" display showing both frequencies with separate transmit and receive frequencies*. If the numbers in this display are **grey** the "Frequency Lock" (*configured in the menu)* is active. If this display is **yellow**, a transverter offset has been configured.

- **RIT+Tuning Display:** Above and to the right of the main frequency display is a smaller display that is offset from the main display if the **RIT** is set to something other than zero.

- **Band Display:** To the right of the main frequency display is an indicator of the amateur band in which the current frequency is tuned with click areas underlined in red and green to increment or decrement the band. If the current frequency is outside an amateur band it will display "Gen" (e.g. "General Coverage"), depending on configuration settings.

- **Mode Indicator:** Above the MHz digits of the main frequency display is the current demodulation mode displayed on a blue background.

- **Zones**: These three zones are controlled via clicking the top box to toggle through the controls for each zone, and then swiping up and down in the three LCD screen regions, left-side, middle, right-side. Click the DSP button to enable access, and then swipe in the regions outside of the DSP virtual keypad that is displayed.

Along the top there are a number of additional indicators:

- **Time & Date:** This can be set by clicking MENU, then opening the configuration menu. Please refer to the instructions on page 5, and the linked instruction video.

- **S-Meter:** This S-meter is nominally calibrated so that S-9 equals 50 microvolts into a 50 ohm load with each S-unit representing 6 dB. Practically speaking, the usable range of the S-meter ranges from about S-3 to something a bit higher than “40 over” which, if you were “run the numbers” about matches the dynamic range of the receiver! The bottom half of the S-Meter's graticule (“S0-S9”) is normally white in color, but if the receiver's A/D converter experiences an overload condition, it will turn red. On bands with strong signals it is normal for this to momentarily flash red as the internal gain control adjusts itself. In **Figures 2 and 3** the S-meter is displaying a signal level of S-9.

- **PO:** The S-Meter scale, when in transmit mode, also indicates the output power from the transmitter.

- **Multi-function display:** Below the S-Meter and Power Output meter is a multi-function meter that, by clicking, may be used to select one of three modes: **SWR**, **AUDIO**, and **ALC**.

  - **SWR:** When in transmit mode, this meter indicates the calculated VSWR. *Note that the VSWR is calculated only when the forward power exceeds 0.25 watts. When in SSB mode, this indicator will not show any VSWR indication unless/until there has been some RF power that exceeds the minimum power, allowing a calculation to be made.*
  - **AUDIO:** This indicates, in dB, the relative audio level being applied to the MIC/Line input.
  - **ALC:** This indicates, in dB, the amount of gain ***reduction*** that the ALC is applying while in transmit mode. 3-12dB of indication during typical speech is normal.

- **VCC:** Below the Signal meter is a voltmeter that indicates the current supply voltage. Below 4.40 volts, the digits are displayed in red, orange below 4.20 volts and yellow below 4.00 volts. The receiver may function as low as 3.3 volts.

- **TRACE:** To the right of the Mode button. Clicking trace will enable or disable the trace outline shown on the scope display.

- **Freq. Step [<< >>]:** Just below the DSP button, and slightly to the right. If clicked once the frequency will step + or – 1KHz. A long press will keep stepping the frequency in 100Hz increments, ideal for fine tuning. Clicking the lower 3 digits of the main frequency display will set the frequency to 0 Khz or 0.5 Khz.

- **DSP Setting:** Middle far left is the indicator of the DSP mode button. The modes available are: **"OFF"**, **"NR"** *(Noise Reduction)***, "NOTCH"**, and **"NR+NOT"** *(Noise Reduction and Notch)*. Trace may be automatically disabled in the CPU overloads.

Spectrum display:

Below the frequency readout, shown in **Figure 2** and **Figure 3** is a spectrum display that shows signals that are on either side of the current tuned frequency. Along the bottom of the spectrum display is a frequency scale that shows the frequency scaling of the graticules rounded to the nearest kHz.

This display works very much like a spectrum analyzer with the vertical scale being represented logarithmically, the number of dB/division being selectable by the user. To further the analogy to a spectrum analyzer, the "reference level" *(the signal level at which a particular strength is indicated)* is automatically adjusted via an AGC *(Automatic Gain Control)* within the spectrum scope that operates independently from the receiver's AGC that automatically scales the strongest signal within the passband such that it is at/near the top of the scope – this, to allow the representation of widely varying signals on different bands without the need of user adjustment.

Waterfall display

Figure 4 shows an alternate method of displaying signals near the currently-tuned receiver signal is the **Waterfall Display**. In this mode, the frequency is displayed along the "X" (horizontal) axis, just as in the case of the Spectrum Scope but instead of the signal strength being displayed as height, it is displayed as relative "brightness". The waterfall displayed is so-called because it can convey the history of recent signal in time by showing the most recent signals at the bottom, but when new signals are analyzed, the older signals are displaced vertically and the newest signals are placed along the bottom. In this way, one has a quick visual "history" of what has occurred not only on the center frequency.

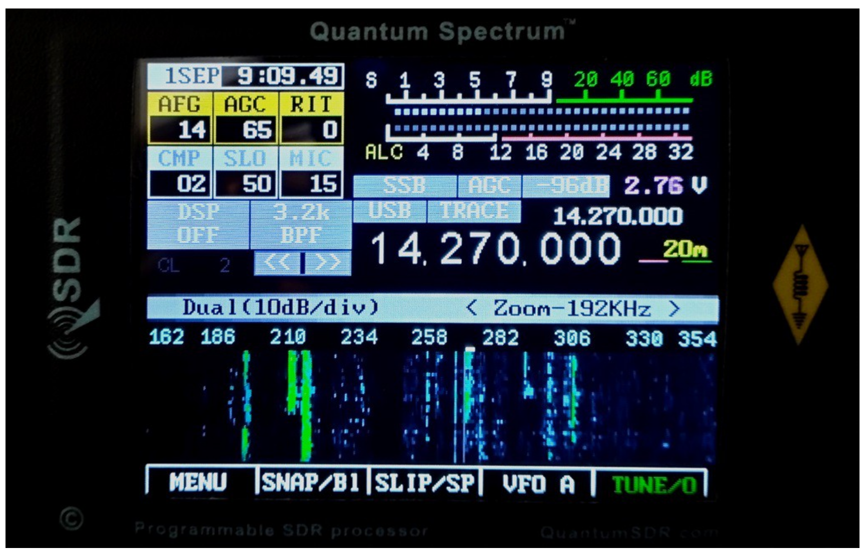


Figure 4: The waterfall display unzoomed showing +/- 96 kHz (192 kHz) of a band segment. Zoom mode allows 96 to 3 Khz to be viewed.

Scope display

Figure 5 shows an alternate method of displaying signals near the currently-tuned receiver signal. This is the Scope Display.

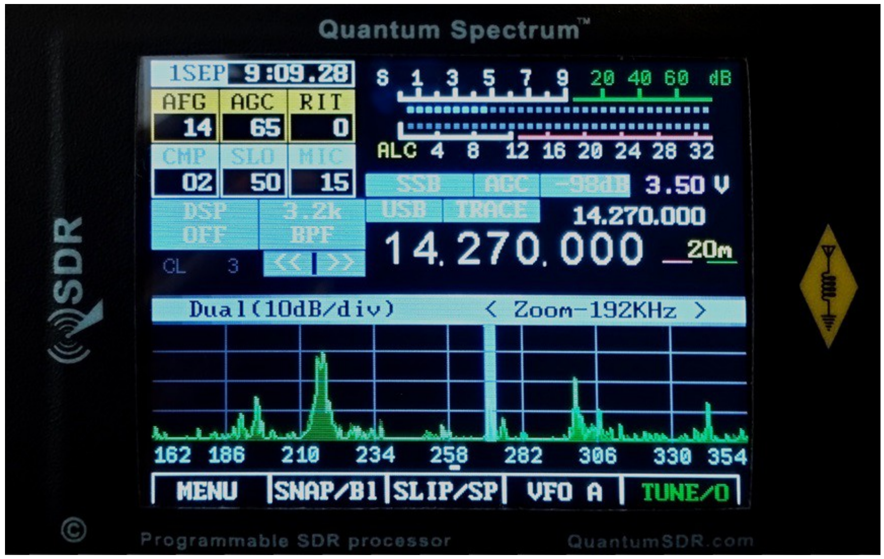


Figure 5: The unzoomed scope display showing +/- 96 kHz (192 kHz) of a band segment. Zooming to 3KHz is permitted.

By default the scope display shows all RF levels across the spectrum, but making a long-press on [SNAP/BI] the spectrum can be balanced to show more defined signal peaks. This is best done either with the antenna grounded or disconnected, on a quite area of the band, or fast tuning the transceiver while balancing, which takes a few seconds to complete.

Displaying both Spectrum and Waterfall:

Click the header area shown in light blue on the far left-side, to the left of “Dual(19..)”. This will cycles through the available display modes-

Options available to both the Spectrum Scope & Waterfall Display:

An adjustable “smoothing” filter (menu item “Scope/Wfall Filter”) is available that dramatically improves the visibility of rapidly-changing signals In the menu system, the range of the Spectrum Scope may be set to span either +/- 96 kHz or +/- 3 kHz, with the scope's AGC operating only on signals within the displayed span – see the menu setting “Spectrum magnify” for more information.

Also available are “Window function” selections that operate on the input FFT data to both the Spectrum Scope and the Waterfall Display that pre-process the spectral data to minimize “spill-over” of adjacent FFT “bins”. What this can do is make the Spectrum Scope and Waterfall display look “sharper” and prevent a strong signal from “leaking” over and covering a weak one.

Connectors:

On the right-hand side of the transceiver on the UI board are four 3.5mm three-conductor connectors. Starting from the top, these connectors are:

- **Headphones.** This is the receive audio output. It will work with headphones earbuds or 8 ohm speakers. An amplifier is available if required. **WARNING**: Ensure the volume level is set below AFG=10 before inserting earbuds or headphones

- **CAT.** This connector is for connecting the TTL serial port of the associated transceiver. Elecraft/Kenwood, USDX, ICOM and Yaseu protocols are supported.

- **EXT.** This jack has the DIT (PA TX) and DAH signals as shown in the connection diagram at the end of this manual. When TX is active, the data will show in red, and the audio will be muted.

- **5V DC.** This is the power connector. The outer shell is negative and the inner conductor is positive. **WARNING:** Never reverse connect or connect to a higher voltage and the unit will be destroyed.

- **Debug.** This connects to *a computer to allow firmware updating. Please contact us for instructions.*

- **IQ.** This is used for interfacing with the receivers IQ signal, available either through an IQ port or tap to the circuit board, as provided by for example, the Electraft KX2.

- **PC.** This port connects to the PC to support IQ streaming to the PC, and provide a CAT communications bridge to the PC using the Yaseu FT-819 protocol.

RESET BUTTON:

On the rear of the unit there is a 5mm hole, where the stylus can be used to gently activate an MCU reset.

Operational modes and functions:

Receive mode:

After powering up, the Spectrum DSP will revert to receive mode on the last frequency, in the mode and using the audio bandpass filter that was in use when it was last powered down ***using the POWER button***.

To adjust the volume click the DSP mode button and make short swipes up and down on the left-side of the screen. The AFG box number will change as this is done, adjusting the volume level.

Transmit mode:

In transmit mode most of the controls are locked, this being done to prevent the change of frequency, filter type and mode during mid-transmission.

TUNE mode: Virtual dial

Tune mode may be entered by pressing the button located below the **TUNE/O** button on the lower screen.

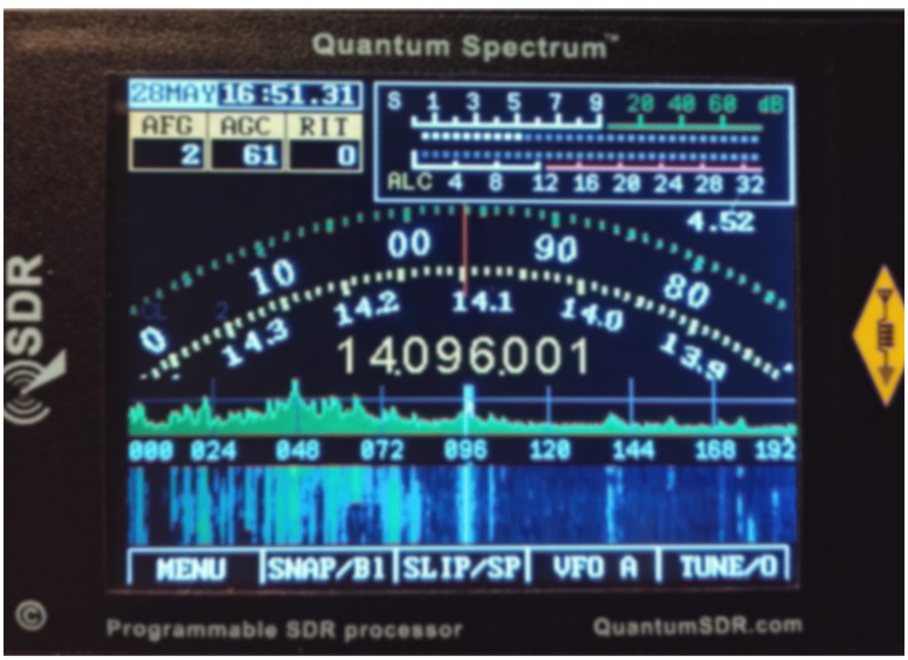


Figure 6: The virtual dial mode provides accurate frequency control.

Draw a circle on the screen as if you were spinning a dial to tune. If you want to tune faster you can also make a long-press on [VFO A] which will then turn red to indicate that fast tuning mode is active. Both modes provide accurate frequency control.

Pressing the **TUNE** button again will exit to the scope tuning mode. Clicking anywhere on the scope will change frequency if the Spectrum is in Master or BiSync modes and the CAT cable is connected to the transceiver.

VFO A *(or VFO B)*:

When not in Menu mode, the [VFO A/B] button toggles which VFO, A or B, is currently in use. This display will change, always indicating the currently-active VFO.

If **SPLIT** mode is *not* active, the currently active VFO's frequency, filter selection and mode are used for both receive and transmit.

SPLIT mode:

To activate **SPLIT** mode make a long-press on the [SLIP/SP] button. When active the current VFO's frequency and filter are used for receive while the "other" VFO's frequency is that used for transmit: The transmit mode is ***always*** that of the "active" (receive) VFO. *The SPLIT mode will be discussed in more detail below.*

When SPLIT mode is on, the radio uses the currently-selected VFO's mode for **<u>both</u>** receive ***and*** transmit, the current VFO's filter and frequency for receive and the "other" VFO's frequency for transmit. In this mode, the main frequency display is also changed, showing both the receive ***and*** transmit frequency, separately.

To set up for SPLIT mode one might do the following: (transceiver dependent function)

- Activate the SPLIT function by long-pressing [SLIP/SP]. The display now shows two frequencies.
- Suppose that a DX station is transmitting on 14.155 and receiving on 14.165, USB. In that case, ***you*** would transmit on 14.165 and receive on 14.155.
- Tune to your transmit frequency of 14.165 MHz – the *receive* frequency of the DX station.
- Press the **VFO A/B** button to move that frequency to the "other" VFO: That is now your transmit frequency.
- Tune to your receive frequency of 14.155 – the *transmit* frequency of the DX station – and also set USB mode and your desired filter bandwidth.
- ***You are now ready to go!*** - *It doesn't matter which frequency is in VFO A or B.*

Important Comments related to SPLIT mode and VFO A/B:

- The SPLIT mode works ***only*** on the same band – this to prevent the destructive battering of the band-switch relays that might occur with crossband operation – which would also slow down transmit/receive switching.
  - Note that it is possible for one to set the STEP to 100 kHz and using the main knob to tune the receive frequency to another band and operate split that way, but this is ***not*** recommended and you do this at your own risk!
- When using FM mode it is possible to use the SPLIT function for repeater operation if one VFO contains the repeater input frequency and the other contains the repeater output frequency. By "swapping" the VFOs one may also do a "reverse" function and listen to signals on the repeater's input frequency.

Menu button operation:

In “normal” operation the spectrum display will be visible on the screen and the five “Function” buttons along the bottom of the display will have the following functions:

- **MENU** (button **F1**) – This enters the menu system, allowing the configuration of the transceiver. Pressing and holding this button will save all settings to EEPROM.
- **METER** (button **F2**) – This button selects the mode of bar graph below the S-meter which is used to display different parameters while transmitting. Repeatedly pressing this button selects, in turn, the display of SWR, AUD and ALC.
- **SPLIT** (button **F3**) – This button toggles “SPLIT” mode on/off. When on *(“SPLIT” is yellow)*, the transmit and receive frequencies are separated using VFO A and B as shown on the main frequency display.
- **VFO A** or **VFO B** (button **F4**) – This button toggles whether VFO A or VFO B is the “primary” VFO. The VFO that is being displayed is ***ALWAYS*** the one being used for receive.
- **TUNE** (button **F5**) – This button toggles the **TUNE** mode on/off. Pressing and holding this button will disable transmit as indicated by this indicator being displayed in gray.

In MENU mode:

Pressing the **MENU** button *(e.g. button **F1**)* will enter the main menu system by which many parameters of the transceiver may be configured: These parameters will be discussed in detail later. Pressing-and-holding this button will save all settings to EEPROM.

Upon entering the **MENU** mode several of the “soft” buttons along the bottom of the screen will change their function:

- **EXIT/S** (button **F1**) – This exits the menu system, returning to the main display. Pressing-and-holding this button will save all settings to EEPROM.

- **PREV** (button **F2**) – This button goes backwards one screen or six menu items. Pressing-and-holding this button will jump to the beginning of the menu, or to the end of the menu if already at the beginning.

- **NEXT** (button **F3**) – This button goes forwards one screen or six menu items. Pressing-and-holding this button will jump to the end of the menu, or to the beginning of the menu if already at the end.

- **DEFLT** (button **F4**) – This button resets the currently-selected item to its default setting.

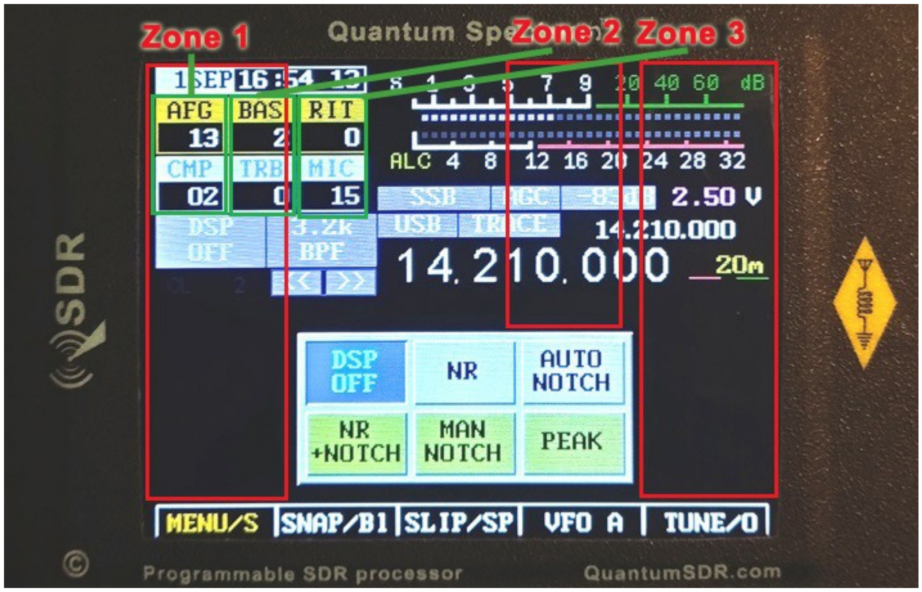
- **PWR** (button **F5**) – shutdown, immediate press Confirm to complete.

Note: If an item has been changed in the menu system that may need to be saved to EEPROM by long-pressing the EXIT/S button.

How to adjust values and options using the touch screen:

IMPORTANT SECTION!
All settings are adjustable through 3 zones, these can be seen in the image below:

Zone 1: See the **AFG** column.
Zone 2: See the **BAS** column.
Zone 3: See the **AFG** column.



As modes are changed these 3 columns show different names and values, but the operation stays exactly the same.

To **Activate** the adjustment mode, press the **[DSP]** button, shown above in the image as DSP with OFF below. The DSP virtual keypad will then be displayed, as shown above.

Next we click on the top the the column that we want to change. For example, to change the volume, AFG must be shown in the top box of the left column. If you click the AFG box, the yellow highlight will change to the box below, allowing that variable to be adjusted.

So the left column, Zone 1, can now be adjusted by stroking up and down on the left-side of the LCD in the **red Zone 1 region**. With AFG highlighted and the DSP menu displayed, short up or down swipes in this region will change the volume level, and the AFG number will change. When finished just click the DSP button again to clear the DSP keypad.

This applies equally to the other two columns. Clicking the top box of each column will cycle the boxes through all the different settings, much like how a touch interface on a cell phone operates.

For example, click the top box of column 2 (it may show AGC on your screen) and the text shown will cycles from AGC to SLO, BASE and TREB. Whatever has the yellow highlight can be adjusted by swiping up or down in its Zone area shown in red, Zone 2 in the case of column 2. With a little practice it becomes easy to use, and there is a video available on YouTube here that shows how to use the LCD interface.

Configurable options on the main touch screen:

In the upper left corner there are a number of items on the main screen that are configurable using the buttons and/or encoders.

- **AFG** - "AF Gain" (Volume Control). This is used to adjust the audio level feeding the speaker/headphone jack. Clicking the AFG box will select whether **AFG** or **CMP are adjusted** *(see below)* with the "un-selected" item being "grayed" out. **AFG** is ***always*** enabled when in **Menu** mode.

- **STG** - "Sidetone Gain" ***while in CW mode***. This is used to adjust the level of the sidetone that is heard during keying while in CW mode and while in TUNE mode by swiping up/dwn after pressing the DSP button, on the left-side of the screen. Press the AFG box to select whether this encoder adjusts **STG** or **AFG** with the "un-selected" item being "grayed" out. Sidetone Gain is also adjustable from the main menu. ***When <u>not</u> in CW mode this is replaced with "CMP".***

- **CMP** - "TX Compression Level" ***while <u>not</u> in CW mode.*** This is used to adjust the amount of audio compression when in voice mode. ***When in CW mode this is replaced with "STG"***.

- **RFG** - "RF Gain". This control, as the setting is decreased, causes an increased deflection in the S-Meter and a commensurate decrease in the receiver sensitivity. This functions in exactly the same way as the "RF Gain" control on a traditional analog receiver and is typically used to limit the receiver sensitivity on a noisy band. the AGC box may be used to select whether this encoder adjusts **RFG** or **NB** *(see below)* with the "un-selected" item being "grayed" out. This parameter may also be adjusted from the main menu.

- **DSP** – This adjust the "strength" of the DSP noise reduction, when enabled. Pressing-and-holding the AGC box will select between this parameter or "**NB**" *(Noise Blanker adjust)* being visible. Turning the DSP on and off will also reset the DSP noise reduction/notch engine.

- **NB** - "Noise Blanker". This control adjusts the "strength" of the noise blanker, with "0" being "disabled." This is a "pulse" type noise blanker operating on the wideband input prior to filtering in the DSP input. As the noise blanker strength is increased, the color of the number changes to warn the user that the higher numbers are more likely to cause degradation of the receive audio. the AGC box may be used to select whether this encoder adjusts **NB** or **RFG** with the "un-selected" item being "grayed" out. Pressing-and-holding the AGC box will select between this parameter or "**DSP**" being visible.

- **RIT** - "Receive Incremental Tuning". This offsets the receiver, in 20 Hz steps, to allow the transmit frequency to be different from that of the receiver with the actual receive frequency being shown on the "sub" frequency display above and to the right of the main frequency display. Button **M3** may be used to select whether this encoder adjusts **RIT** or **WPM** *(see below)* with the "un-selected" item being "grayed" out.

- **WPM** - "Words Per Minute" ***while in CW mode***. This adjusts the Morse sending rate in "Words Per Minute" when using Iambic mode keying. Button **M3** may be used to select whether this encoder adjusts **WPM** or **RIT** *(see below)* with the "un-selected" item being "grayed" out. The Morse WPM setting may is also adjustable from the main menu. ***When <u>not</u> in CW mode this is replaced with "MIC" or "LIN"***.

- **MIC** or **LIN -** "Microphone Gain" or "Line Input Gain" ***when <u>not</u> in CW mode***. This adjusts the Microphone (or Line Input) gain, depending on which is enabled. ***When in CW mode this is replaced with "WPM"***. Pressing-and-holding button **M3** will select Microphone or Line-Input modes. ***Note that if this is changed during transmitting, one must briefly unkey for the change of inputs to take effect.***

Automatic switching of on-screen items when going from receive to transmit:

Using the item in the "Configuration Menu" labeled "**O/S Menu SW on TX**" and setting it to **ON** several of the on-screen items will change automatically when going from receive to transmit and back again when returning to receive when in SSB mode: ***This function is NOT available in CW mode.*** These parameters include:

- **CMP** (in voice mode)

- **MIC** or **LIN** (in voice mode)

This automatic switching facilitates the adjustment of the relevant parameters when in transmit mode without having to pause and press the **M1** and/or **M3** buttons to switch the functions of the relevant knobs.

Note that if you already had selected an alternate function while in receive *(e.g. "**CMP**")* it will "remember" and return to that setting after you have been in transmit and again turned to receive.

Setting the parameter " **O/S Menu SW on TX**" to **OFF** prevents the above parameters from changing when going between receive and transmit.

DSP (Digital Signal Processing) Noise Reduction and Automatic Notch Filter:

the AGC box is used to enable/disable the DSP function, providing the following settings:

- **OFF –** DSP Functions are turned off

- **NR** – Noise Reduction only

- **NOTCH** – Automatic Notch Filter only

- **NR+NOT** – Noise Reduction and Notch Filter

Pressing-and-holding the AGC box will “save” the currently-selected DSP mode, if on, and turn it off. Pressing-and-holding this button again will restore the mode(s) that had been configured when it had been turned off.

The “strength” of this filter may be adjusted using the menu item #10, “DSP NR Strength” - but be very careful with this as it easy to go overboard with this setting. If it is set too high, the artifacts caused by the noise reduction *(e.g. “hollow” or “watery” sound)* can be *worse* than the interference than you are trying to remove!

The “strength” may also be set using the AGC box and swipping up/dwn in the middle-right area without having to enter the menu system. To do this:

- Enable DSP “NR” mode by pressing the **DSP** button and pressing the **NR** button.

- Press the AGC box so that the highlighting changes from **RFG** to **DSP** on the screen.
    - If “**NB**” is displayed instead, press-and-hold the AGC box to change it.

- With **DSP** highlighted, swipe up/dwn in the midle-right of the LCD to adjust the DSP noise reduction “strength”.

- You will note that the number denoting DSP “strength” is greyed out when DSP is turned off and cannot be adjusted.

IMPORTANT OPERATIONAL NOTES related to DSP and the noise blanker:

- All DSP functions are disabled until a few seconds after the radio boots up.

- The notch filter is automatically turned off in CW mode. It cannot be selected when in CW mode. The reason for this is that the notch filter would “kill” CW signals!

- ALWAYS turn all DSP modes off when you are using any “sound card” *(digital)* modes such as PSK31, RTTY, SSTV, etc. ***DSP is <u>NOT</u> compatible with these modes!***

- The noise blanker is always disabled in the wide bandwidth *(5, 6, 7.5 or 10 kHz)* mode.

- The noise blanker is disabled in AM mode.

- Enabling the noise blanker *and* DSP can cause the user interface of the Spectrum DSP to slow down significantly! What this means is that the response to button-presses and the updates of the spectrum scope can be significantly slower. *(You have been warned!)*

There are additional “advanced” configuration settings related to the DSP modes available: See the items in the menu system and the section on “Advanced DSP Settings” later in this manual.

Tips to minimize processor loading when using DSP:

- The DSP Noise Reduction and the Automatic Notch Filter ("Notch") are ***separate*** functions that operate independently. Because of this, operating on "**NR+NOT**" mode takes more processor "horsepower" than either "**NR**" or "**NOTCH**" alone.

- The noise blanker takes about as much processor power as both the DSP NR ***and*** "Notch" put together which is why turning on the noise blanker in addition to DSP can significantly slow down the transceiver's response – and also why the noise blanker is disabled in AM mode and when set to a wide bandwidth mode – either of which take more processor power in their own right!

- When DSP NR is active, the parameter "**DSP NR FFT NumTaps**" can significantly change processor loading: The higher this value, the more loading. If you need to have DSP turned on, but you find the user interface to be operating too slowly, try setting this to a lower value: This can decrease the "quality" of the noise reduction somewhat, but it will free some processor power.

WARNING:

- **It is possible to select the combination of wide bandwidth, DSP noise reduction and DSP Notch (e.g. "NR+NOT") while in AM mode. This combination can "stall" the radio with too much processor power, making operation sluggish and result in distorted audio. If you do this, you may press-and-hold the DSP button to disable DSP and "un-select" some of these options.**

Using the Spectrum DSP

A brief overview:

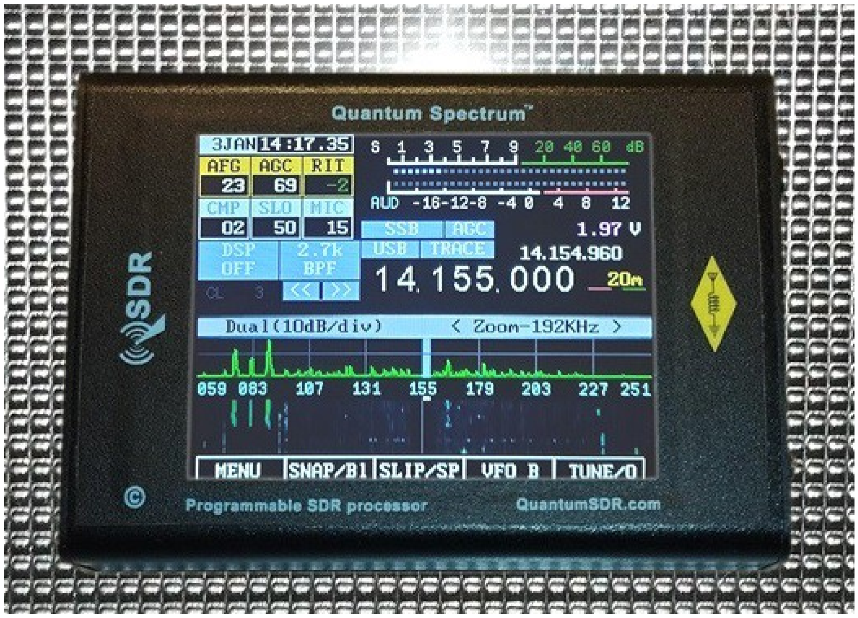


***Figure** 7: Front panel of the Spectrum DSP*

NOTE: If you are unfamiliar with the Spectrum DSP, please refer to the section of the manual: "**Before you get on the air - Initial set-up of the Spectrum DSP"**

To turn on the transceiver, press the **POWER** button briefly and the display should light up, go through its attribution and boot-up screen and display the frequency and spectrum display.

Receive:

- Adjust volume clicking the **DSP** control and sliding up and down on the LCD left-side.
- Tune the frequency by clicking on the Spectrum scope, and dragging the points left and right from the tune bar to adjust, or using the **<< >>** buttons. *Long-press on the <<>> buttons will auto step by 100Hz in either direction.*
- Change the band by clicking on the MHz value of the main frequency display and selecting the required band.
- Change the mode *(USB, LSB, CW, etc.)* using the Mode button**.** *Note: Pressing-and-holding this button will force the selection of "disabled" modes.*
- The Filter button selects the receiver bandwidth. *Note: Pressing-and-holding this button will force the selection of "disabled" bandwidths.*
- Pressing the **DSP** button will display the available DSP noise reduction modes.
  - **Pressing and holding** the **DSP** button will disable the DSP, saving the current settings while pressing and holding again will restore the last-used mode. The available DSP modes are:
    - **NR** = Noise reduction only
    - **NOTCH** = Automatic notch (tone) filter only
    - **NR+NOT** = Both Noise reduction ***and*** Automatic notch filter.

There are certain configurations where some/all of the DSP functions are not available. For example, the notch filter is disabled in CW mode *(for obvious reasons!)* and DSP is completely disabled in FM mode.

- If **RIT** is desired, use **RIT** to shift the receive frequency: The ***small*** frequency display will show **actual** receive frequency display when **RIT** is set to non-zero, but the large display will show the **transmit** frequency.

Transmit: (set-up dependent)

Set the receive frequency and mode, setting the desired output power by pressing the **RIT** box. ***Note that it is recommended that for voice modes that "full" power <u>not</u> be used unless you have carefully configured for clean, linear output power.***

Initial SSB transmit audio set-up: (set-up dependent)

- Preferably, connect the Spectrum DSP to a 50 ohm dummy load capable of handling at least 10 watts. Alternatively, you may tune to a **<u>clear</u>** frequency while connected to an antenna with a ***known-good*** 50 ohm match.
- Use the mode button to select LSB or USB mode as desired.
- Press the **AGC** box to select the **AUDIO** meter.
- Connect the microphone to connector **(optional port)**: The Spectrum DSP is typically used with an electret-type microphone element and power for the microphone element is supplied by the radio.
- Press the **RIT** box to switch from **RIT** to **MIC**. If the box to the right-hand side of **RIT** shows "**LIN"** which indicates that line-input mode is active, press the **RIT** box to change it to **MIC.** Press the **RIT** box as necessary to highlight **MIC** on the display: This allows the adjustment of the microphone gain.
- Now, key the radio using the Push-to-Talk (PTT) button on the microphone: The spectrum display should freeze.
- **Speak normally** into the microphone. You should see the indicator on the **AUDIO** meter bounce upwards. While speaking, swipe up/dwn th the right side of the LCD to adjust the **MIC** parameter, so that the **AUDIO** meter indication peaks up to +4 or so *(in the red)* on peaks. Occasional, higher, higher peaks are permissible, but avoid settings that cause full-scale indications which could imply distortion.
- Release the PTT button and press the AGC box to select the **ALC** meter.
- Press the **AFG** box to highlight the **CMP** on the display: This will allow the adjustment of the compression level of the speech processor.
- Press the PTT button and speak normally again. You should see the **ALC** meter indicate upwards on voice peaks occasionally: If it does not, increase the **MIC**rophone gain slightly.
- Adjusting **CMP** (set-up dependent) to a higher value will increase the aggressiveness of the speech processor: A value of **2** for mildly compressed, a modest value and a value of 12, while very "punchy" can be used to maximize "talk power" will sound very "processed" and is likely to be unpleasant for normal, casual QSOs. *The value of "SV" will select custom settings – see the menu for additional information.*
- Once you have configured the settings to your satisfaction, press-and-hold button **F1** to store them in memory.

What to do if you notice that the ALC or AUDIO meters jump when you push PTT:

In a quiet room with an antenna or dummy load connected to the Spectrum DSP, set the **METER** mode to **ALC** and key the microphone/transmitter without talking and note if **ALC** meter jumps at the instant that you key the transmitter and goes down again. Next, switch the **METER** mode to **AUD** and key the microphone/transmitter again, watching the **AUD**io meter.

If you notice that either meter jumps upwards when you key the transmitter and drops down again your keying the transmitter may be causing either an electronic “click” or mechanical “clunk”, “de-sensing” the transmitter's ALC. This can be caused by the the powering-up of the electret element in the microphone when the radio is keyed and/or by the (noisy!) mechanical action of the switch – but the result can be the same in either case: A temporary “desense” when you start talking and/or an annoying sound heard by the station receiving you!

To minimize this adjust menu item **“TX Mute Delay”** which will keep the microphone audio muted for a short period after keying up. The parameters are adjustable from 0 (off) to 25, which keeps the audio muted for a full 250 milliseconds (one-quarter of a second) after the microphone is keyed.

It is recommended that one finds the minimum value to reliably suppress the appearance of the microphone key-up noise and then increase it by 50%.

Comments when using AM:

- AM transmission operates the same way as SSB, but ***frequency translation mode <u>must</u> be activated.*** Remember also that the unmodulated carrier in AM will be ¼ that of the PEP power in SSB!

Important information regarding the “Frequency Translate” mode:

Menu item “**RX/TX Freq Xlate**” selects the enabling/disabling of baseband frequency translation in the receiver/transmitter. When the translation is active, instead of the receiver operating at and around "DC", the signals are mathematically shifted from 6 kHz (above or below – user-selectable). Whether or not frequency translate mode is enabled is displayed on the start-up splash screen.

Performing this frequency shift can help forgive a lot of the "sins" that occur with "DC" conversions - the most obvious of which are that ANY noises in the power supply as well as the 1/F noises of op amps, mixers, A/D converters and the like tend to show right up in the received audio. With the signals at microvolt levels, it is a ***real fight*** to minimize these signals! These signals/problems can show up as:

- Hum
- Howling
- Audio feedback, particularly at higher volumes

- Buzzing with the dimming of the backlight
- Noises from the I2C communications (e.g. “ticking”)

It should be noted that these code modifications ***DO NOT*** relieve the builder of the ***strong recommendation*** that one perform the modifications in the "Spectrum DSP Board Modifications" file, particularly the U3a and MCU and LCD power supply modifications (for UI board 0.3) but they should go a long way toward reducing the artifacts that can still occur even after making those modifications - even to the point of gaining an extra S-unit or two in sensitivity.

Menu item “**RX/TX Freq Xlate**” has the following options:

- **OFF** - This is the original operation of the transceiver with the receive (and transmit) signals operating at and around zero Hz.
- **RX LO HIGH** - In this mode the signals are shifted BELOW zero Hz by 6 kHz, requiring that the local oscillator be shifted up by the same amount. The received signals are tuned at the first graticule left of center on the spectrum scope.
- **RX LO LOW** - In this mode the signals are shifted ABOVE zero Hz by 6 kHz, requiring that the local oscillator be shifted down by the same amount. The received signals are tuned at the first graticule right of center on the spectrum scope.

For various reasons (e.g. the use of USB on higher bands where the potential for zero-HZ interference is highest) **the use of “RX LO LOW” is recommended for best performance!**

Quirks and side-effects:

When the translate mode is activated and “magnify” mode is not turned on you will note that the receive signal is ***no longer in the center of the spectrum scope!*** Along the bottom of the spectrum scope you'll observe that the frequency display is changed, with the frequency in kHz being displayed in full under the graticule, being shifted left or right as noted above.

If you have used other SDR software – particularly “sound card” SDR rigs on computers – you will already be familiar with this sort of shift!

Using the Spectrum DSP with computer "Sound Card" *(e.g. digital)* modes via the Line-Input and Line-Output connections:

The Spectrum DSP may be connected to a computer, tablet or smart phone via audio cables and the PTT line on the Microphone cable to allow modes such as SSTV, PSK31, WSPR or other digital "Sound Card" mode. To do this, configure the transceiver as follows:

- Using the mode button select **USB** mode: All digital modes are operated using USB, ***regardless of band.*** In this way the audio frequency of the digital signal may be added to the frequency display to calculate the ***actual*** transmit/receive frequency.
- Set **RIT** to zero by swiping up/dwn in the right region with the DSP menu showing (which activates swiping changes)**:** Press the **RIT** box as necessary until **RIT** is highlighted to allow adjustment. When using a digital mode the RIT ***MUST*** be disabled or else you will have difficulty making contacts!
- Set **CMP** to zero by swiping in the left-side of the LCD with the DSP menu active**:** Press the **AFG** box as necessary to highlight **CMP** to allow adjustment by swiping in the left-side of the LCD. When using a digital mode, the audio compressor must be set to ***MINIMUM*** (0) or else it may degrade the digital signal!
- Before connecting the external device *(Computer, tablet, phone)* set the audio output level to mid-scale. Also set the audio input gain to approximately mid-scale as well.
- For receive, one may use any of the available receive audio filters, but it is recommended that the Wide filter ***not*** be used! If narrow (300Hz, 500Hz or 1.8 kHz) filters are used, one may shift the center frequency of that filter in the menu to suit the passband for that mode, but be aware that it is possible to run ***too narrow*** a filter for some of the "wider" digital modes! In the vast majority of cases the 2.3kHz filter will be adequate.
- **Be certain that DSP filtering is turned off!** The DSP noise reduction or notching on *any* radio necessarily alters signals and doing so can degrade them, making them difficult for the attached computer/device to decode!

Connect the Line-Input jack (optional port) of the Spectrum DSP to the audio output of the device you are using to generate the audio and connect the Headphones jack of the Spectrum DSP to the audio input of that same device.

To key the transceiver, you will need also to connect a cable to the EXT to trigger TX mode by grounding the line: Typical rig-computer interfaces will easily accommodate this connection.

- Preferably, connect the transmitter to a 50 ohm dummy load capable of handling at least 10 watts. Alternatively, you may tune to a **<u>clear</u>** frequency while connected to an antenna with a ***known-good*** 50 ohm match.
- Press the **RIT** box to select **LIN** mode. You may need to press-and-hold this button to change from **MIC** to **LIN**. Press the **RIT box** as necessary to highlight **LIN**.
- Press the **AGC** box to select the **AUDIO** meter.
- Using the program running on the external device, key the computer using the selected mode. If the program has a "test" mode, use it for this.
- Adjust the **LIN** setting by swiping up/dwn in the right-region for a reading on the **AUDIO** meter of +2 to +4.
- ***Make sure that you have set "CMP" to 0 as noted above!***
- Un-key the transceiver.

- Make a note of the settings that you have used for future reference.
- Find a signal on the bands representative of the mode and adjust the audio input level of the external device for approximately “mid-scale”.
- It should be noted the the **Headphone** jack will contain the ***transmit*** audio. This is an artifact of the hardware configuration.
- Once you have configured the settings to your satisfaction, press-and-hold button **MENU** then **PWR** and **Confirm** to store them in memory.

TUNE mode: (set-up dependent)

The **TUNE** button may be used to send an unmodulated (CW) carrier for brief testing, such as checking the RF power output or the VSWR/matching. The **TUNE** function is also used for initial adjustment of various parameters *(TX Gain, Phase)* as described elsewhere in detail.

The operation of the **TUNE** mode is very simple:

- Press the **TUNE** button: Transmit is enabled and the indicator turns red.
- Press the **TUNE** button again: The Spectrum DSP stops transmitting and the indicator turns white.

Comments about the TUNE mode:

- When set to CW mode, when **TUNE** is activated the Spectrum DSP will produce a carrier ***above*** the dial frequency by the amount of the setting of the **“CW Side/Off Freq”** (e.g. sidetone frequency).
- When set to SSB mode, when **TUNE** is activated the Spectrum DSP will produce a carrier that is offset from the dial frequency by 750 Hz – the same as the audible sidetone. This carrier will be below the dial frequency in LSB mode and above it in USB mode.
    - **Note:** There will be no audible sidetone in “SSB TUNE” mode when Frequency Translation is enabled.
- Pressing-and-holding the **TUNE** button will toggle the **TRANSMIT DISABLE** function. If this mode is on, the **TUNE** indicator will turn grey and all transmit capabilities of the Spectrum DSP will be disabled. *This is the same as the parameter “**Transmit Disable**” in the configuration menu.*
- **TUNE** mode does not function in AM mode.

Configuration of the Spectrum DSP for CW operation:

- Connect a key or paddle to jack **J2** on the RF board: This is the connector next to the DC power input.

For connecting a paddle for Iambic keying:

- The **TIP** of the connector is **DIT**.
- The **RING** of the connector is **DAH.**

*Note: The "dit" and "dah" may be swapped using the "**CW Paddle Reverse**" menu setting.*

For connecting a straight key, mechanical semi-automatic key *(e.g. a "bug")* or an external keyer/computer:

- The **RING** of the connector keys the transmitter.

  Note that the DAH/Straight Key connection is the same as the "PTT" line on the Microphone connector.

Now, press the **MENU** button (**F1**) and use the **NEXT** and **PREV** buttons (**F4** and **F3**, respectively) to navigate to the screen containing the menu item "**CW Keyer Mode**", noting the setting to the right of it. The three possible settings are:

- **IAM_A** – Iambic mode "A". Using paddles, alternate dots and dashes are sent with both paddles are depressed, stopping with the last dot or dash that was sent while the appropriate paddle was depressed.
- **IAM_B** – Iambic mode "B". The same as mode "A" except that keying continues by sending one more element – a dot if the paddles were released during a dash and vice-versa.
- **STR_K** – Straight Key. This would be used for a straight key, a "bug"or external keyer/computer.

Additional items on this menu *(you may need to scroll to another screen using **ENC2**)* include:

- **CW Paddle Reverse** – This reverses the DIT and DAH positions of the paddle, affecting ***ONLY*** the IAMBIC modes when using the built-in keyer.
- **CW TX->RX Delay** – This sets the delay, after the last CW element, before the transceiver returns to receive mode.
- **CW Side/Off Freq** – This sets the offset frequency and sidetone in CW operation, adjustable in 10 Hz steps.
  - **Note:** If the sidetone frequency is adjusted, the center frequencies of the 300 Hz and 500 Hz filters should be adjusted to compensate to keep the frequencies within the center of the filter passband!
- The parameters **CW Keyer Speed** and **CW Sidetone Gain** are adjustable from the main display and will be discussed shortly.
- **CW Freq. Offset –** This sets the display/shift mode to be used for CW operation: ***For more details on this parameter, see the MENU section.***

To configure for CW operation:

- Press the mode button to select the CW mode.
- Press the filter button to select the desired receive audio bandwidth if needed.
- Press the RIT box until **WPM** is highlighted, then press the DSP button, and then swipe up/down in the right side of the LCD to set the desired WPM sending speed in words-per-minute. ***This parameter has no effect if set to straight-key mode.***
- Press button the **AFG** box to highlight the **STG** parameter: Ensure the DSP menu is displayed, as this enabled the up/down swiping regions, then swipe up or down on the left-side of the LCD to adjust the **STG** value.
- Press the paddle/key connected to the **EXT** port to cause the Spectrum DSP to enter transmit mode: Press DSP show the DSP pad shows, then swipte up and down on the left side to adjust the volume of the sidetone. *Note that the volume control (**"AFG"**) setting has no effect on the level of the sidetone.*
- *Once you have configured the settings to your satisfaction, press-and-hold button **F1** to store them in memory.*

Miscellaneous notes and tips:

- The DSP "NR" (Noise Reduction) mode may be used to advantage when in CW mode, but note that the DSP "NOTCH" mode is always disabled because it would "kill" CW signals!
- The sidetone frequency is ***exactly*** that of the amount of transmit offset from the dial frequency.
- If the parameter **"CW Side/Off Freq"** is changed - which changes the sidetone/offset frequency - **remember to change the the center frequencies of the 300 Hz and 500 Hz filters** so that the center of your receive filter passband will match your transmit frequency. If you do not do this a station that returns to you *on your frequency* may do so outside the passband of your receive filter!
- There is a slight interaction between the power setting, the perceived loudness of the sidetone gain and the sidetone gain setting. This is a known issue, but it has not been a cause of complaints.
- **NOTE:** Refer to the menu item "**CW TX/RX Offset**" to set up the transceiver for USB, LSB or "Automatic" USB/LSB operation as desired. You may also configure the transceiver so that the frequency displayed is that of the **transmit** carrier frequency ***or*** that of the received signal when its pitched is matched to that of the transceiver's sidetone.
- Pressing-and-holding MENU button when transmitting in CW, LSB or USB mode will generate a tone that is equal in frequency to the CW sidetone and transmit-receive offset. This may be used to "spot" the frequency so that you can transmit on the same frequency as the station with which you are communicating. The loudness of this tone may be adjusted using the "Adjustment Menu" option "**Beep Volume**".

The configuration menu system:

The configuration menu may be entered by pressing the **MENU** button.

When in the menu system, it may be navigated using the following touch strokes:

- **Centre-right up/down swipes –** Selects the individual menu item.
- Right-side up/down swipes – Opens and selects the menu item
- **EXIT/* –** Exits the menu system, returning to the main transceiver display. Pressing-and-holding will save settings to EEPROM.
- **PREV – scrolls the menu up.**
- **NEXT –** Scrolls the menu down
- **DEFLT – Sets the default value for the selected item.**
- PWR – Prepares to shut-down press Confirm immediately to complete.

Important Notes:

- When in **MENU** mode **AFG** (e.g. the volume control.) is always active for left-side up/down swiping.
- Whenever a menu item is changed the warning "Save settings using POWER OFF!" will appear along the bottom of the screen to warn you that any changes that you may have made will ***NOT*** be saved unless you power down the transceiver using the MENU then PWR button.
- If you have made any changes while in the **MENU** system, when you exit the **MENU** system the MENU label will be orange and display "**MENU *"** to warn you that you should save the settings or power down using the **PWR** button to save any changes that you might have made.

There are several separate menus within the menu configuration system:

- The **MAIN** menu. These are the more commonly-adjusted items with the labels in YELLOW.
- The **CONFIGURATION** menu. These are less-frequently adjusted items used for calibrating the radio's hardware with the labels in CYAN *(e.g. light blue.)*
- **Display & Touch** menu**.**
- **CW Mode settings** menu**.**
- **Filter selections** menu**.** Use this to preselect the filters to be available.
- **Touchscreen via menu.**
- **System info** menu.
- **Debug/Exper. Settings** menu.
- **Credits.**

This manual details the key menu settings, with the more obvious menus not needing detailing. If you have any questions or suggestions, please send us an email.

The **CONFIGURATION** menu is hidden unless it is enabled by activating it by setting the last item in the main menu to **ON**.

Note:

All menu items are numbered, but the numbers are omitted here to simplify maintenance of this document as these numbers occasionally change as features are added/modified.

Main Menu configuration items:

Important Note:

If, when the a menu item is changed, it will be necessary to press the **MENU** button then long-press **EXIT/S** to save the changes to the EEPROM.

It is *strongly* recommended that one *NOT* attempt to operate CW when the menu is being displayed! If the menu is being displayed, the CW element timing will be disrupted!

These items are listed in the order that they appear in the menu system.

DSP-related items:

- **DSP NR Strength** – This adjusts the aggressiveness of the DSP noise reduction, with 0 being “weak” and higher numbers correlating to “stronger” DSP noise reduction effects. The relative effects of this parameter are affected by the “advanced” parameters ***– see the “DSP Related Items” section.*** This is the same as the “**DSP**” parameter controlled by **ENC2** on the main screen.

Filter-related items:

- **300Hz Center Freq.** - This sets the center frequency of the 300 Hz CW filter, the options being 500, 550, 600, 650, 700, 750, 800, 850 and 900 Hz. A final option is “Off” which eliminates this filter from the selection when filter button is pressed. The settings will be displayed in white if this filter is currently selected.
- **500Hz Center Freq.** - This sets the center frequency of the 500 Hz CW filter, the options being 550, 650, 750, 850 and 950 Hz. A final option is “Off” which eliminates this filter from the selection when filter button is pressed. The settings will be displayed in white if this filter is currently selected.
- **1.8k Center Freq. -** This sets the center frequency of the 1.8 kHz “narrow” SSB filter, the options being 1125, 1275, 1427, 1575 and 1725 Hz. A final option is “Off” which eliminates this filter from selection when filter button is pressed. The settings will be displayed in white if this filter is currently selected.
- **2.3k Center Freq. -** This sets the center frequency of the 2.3 kHz SSB filter, the options being 1262, 1412, 1562 and 1712 Hz. The settings will be displayed in white if this filter is currently selected. ***This filter cannot be disabled.***

- **3.6k Filter. -** This enables/disables the filter and when set to "Off", this filter will be eliminated from selection when filter button is pressed. The settings will be displayed in white if this filter is currently selected.
- **Wide Filter Select -** This selects the "wide" filter – that is, the next bandwidth above the 3.6 kHz bandwidth with four bandwidth being available: 10 kHz, 7.5 kHz, 6 kHz and 5 kHz. If one of the "AM" items is selected *(e.g. "5kHz AM")* then the selected bandwidth will be available *only* in AM mode but if a "non-AM" item is selected *(e.g. "5kHz")* then this selection will be made available in SSB mode as well.
- **Wide Filt in CW mode –** When ON, the "Wide" SSB filters (3.6 kHz and Wide) will be available for selection when in CW mode.
- **CW Filt in SSB mode –** When ON, the "Narrow" CW filters (300 Hz and 500 Hz) will be available for selection when in SSB mode.
- **AM mode disable -** When ON, the AM mode will be remove from the selection list when the mode button is pressed. *Note that it will still be available if one presses-and-holds the AFG box.*
- **LSB/USB Auto Select** – This enables the automatic selection of LSB or USB, depending on the current band. The available settings are:
    - **OFF** – No automatic selection.
    - **ON** – LSB is selected < 10 MHz, USB is selected >= 10 MHz
    - **USB 60M** – LSB is selected < 10 MHz *except* for 60 meters and USB is selected >= 10 MHz. This setting has been provided for those areas where USB is typically used on 60 meters *(e.g. the U.S.)*

  When "**LSB/USB Auto Select**" is enabled, pressing the AFG box will skip the sideband that is not appropriate for the frequency of operation (e.g. USB will not be selected below 10 MHz) but pressing-and-holding this button when LSB is displayed will change the mode to USB – and pressing-and-holding again will change it back to LSB.

  When "**LSB/USB Auto Select**" is enabled, in order to change to AM you must select a mode ***other*** than LSB (or USB) – such as CW – and then press-and-hold the AFG box: AM will then be selected.

FM mode-related items:

- **FM Mode Enable -** When set to **ON** the FM mode is available with normal presses of the **Mode** button. *Note: Even when off, it may still be accessed if the AFG box is pressed-and-held in CW mode. If the AM mode is also set as being disabled, you may have to press-and-hold **Mode** again to get to FM.*
- **FM Sub Tone Gen –** This selects the frequency of the subaudible tone (in Hz) to be transmitted on FM: The setting of "off" (default) disables the tone. The deviation of this tone is approximately +/-300 Hz in "Narrow" mode and +/-600 Hz in "Wide" mode.
- **FM Sub Tone Det –** This selects the frequency of the subaudible tone (in Hz) to be decoded on FM: The setting of "off" (default) disables the tone decoder so that only "carrier squelch" is used. When this is active it is necessary that ***both*** the squelch be opened **AND** the tone decoded in order for audio to be heard. The "FM" mode indicator on the main display will be backgrounded in red when the selected tone is being decoded.
- **FM Tone Burst** – This enables and selects the transmission of a "tone burst" (a.k.a. "whistle-up") that might be used to activate some repeaters. Two frequencies are

available: 1750 Hz and 2135 Hz. The tone burst is activated by pressing-and-holding button MENU ***while transmitting.***

- **FM RX Bandwidth –** This selects the detection bandwidth when in FM mode: Selection of "audio bandwidth" is disabled as it is irrelevant in FM mode and this setting is unlikely to be changed very often. The selections are:
  - **7.2 kHz** – This is suitable only for "Narrow" operation and even this will result in a bit of added distortion as the filter about as narrow as it may be to pass audio. Using this setting the weak-signal sensitivity is the highest amongst the FM filter bandwidth settings since this narrowest filter also intercepts less noise under weak-signal conditions.
  - **10 kHz** – This is the "default" bandwidth and is suitable for "Narrow" bandwidth and while it will work for "Wide" bandwidth, slight distortion may occur on voice peaks.
  - **12 kHz** – This is wider than necessary for "narrow" bandwidth and recommended for "Wide" bandwidth.
- **FM Deviation** – Two "modes" of FM are available: "Narrow" with +/- 2.5 kHz peak deviation (w/1kHz modulation) and ""Wide" with +/- 5 kHz peak deviation, the former being commonly used on HF and the latter being that which is used on VHF bands except for those instances where the "narrow" (+/- 2.5 kHz) mode is specifically used. These two modes are more or less interoperable, with the following provisions:
  - Operation of "Wide" (+/-5kHz) on "Narrow" frequencies will result in "loud" audio, possible "squelch clamping" and splattering onto adjacent "narrow-spaced" channels.
  - Operation of "Narrow" (+/-2.5kHz) on "Wide" frequencies will result in chronically "low" audio, reduced "copyability" under weak-signal (noisy) conditions and everyone telling you to talk louder!

AGC and other receiver-related items:

- **AGC Mode -** The selections are **SLOW, MED**ium, **FAST, CUSTOM** and **MANUAL**. These related to the "decay" speed (e.g. "hang") of the receive AGC. When in **MANUAL** mode the AGC is disabled and the audio gain is set to maximum – ***see "RF Gain", below.*** **<u>WARNING:</u> Reduce volume level before setting this to MANUAL!**
- **RF Gain -** This is the same as the "**RFG**" (RF Gain) control from the main menu and in this context it is used in conjunction with the **MANUAL** AGC mode.
- **Cust AGC (+=Slower) -** When **AGC Mode** is set to **CUSTOM** this sets the decay rate with a higher setting setting a slower decay. A setting of "12" is equal to the "**MED**" AGC setting. Values lower than 3 are displayed in RED to warn the user that the decay rate of the AGC is likely to be extremely fast, that the resulting audio is likely to be unpleasant and that a bit overshoot/undershoot is possible on the tail end of a signal. This parameter is displayed in orange if **CUSTOM** AGC mode is not selected.
- **RX Codec Gain -** Normally set to **AUTO**, this determines whether or not the A/D input gain on the Codec is automatically controlled based on the input signal levels. If the input levels start to approach full-scale, the gain of the coded is automatically reduced, but if these level have not been attained for a while, the gain is gradually increased again. If this is set to anything other than **AUTO** there is the risk of significantly reducing the dynamic range *(e.g. performance)* of the receiver. When not in **AUTO** mode, the settings range from 8, which is "maximum" gain and the highest susceptibility to overload to 0 which is the lowest receiver sensitivity. *Settings other than **AUTO** are indicated in **RED** to warn the user of likely receiver degradation.*

- **RX NB Setting -** This is the same as the "**NB**" setting on the main screen. This adjusts the "strength" of the noise blanker, with "0" being off.
    - **The noise blanker takes a significant amount of processor horsepower, so some "slowing" of responses should be expected when it is active, particularly if DSP is turned on at the same time!**
    - **The noise blanker is disabled when the menu is displayed, when in AM mode or if a wide bandwidth is selected.**
- **RX/TX Freq Xlate –** This enables the mathematical translation of the receive signals, shifting them from "zero" (e.g. around DC) to + or – 6 kHz. This feature can reduce issues related to direct-conversion receivers such as audio feedback, power supply noise and other noise sources that can degrade receiver performance. The selectable options are:

    - **OFF** - This is the original operation of the transceiver with the receive (and transmit) signals operating at and around zero Hz (e.g. baseband operation.)
    - **RX LO HIGH** - In this mode the signals are shifted ***below*** zero Hz by 6 kHz, requiring that the local oscillator be shifted up by the same amount. The received signals are tuned at the first graticule **left of center** on the spectrum scope.
    - **RX LO LOW** - In this mode the signals are shifted ***above*** zero Hz by 6 kHz, requiring that the local oscillator be shifted down by the same amount. The received signals are tuned at the first graticule **right of center** on the spectrum scope. ***For various reasons (e.g. the use of USB on higher bands where the potential for zero-HZ interference is highest)*** **the use of "RX LO LOW" is recommended for best performance!**

    For more information, refer to the section about Frequency Translation near the end of this document.

Transmit Audio related items: (Set-up dependent)

- **Mic/Line Select -** This selects whether the Microphone or the LINE input is to be used for transmit audio in the SSB mode. This is the same function as pressing-and-holding button **M3** when in a voice mode.
- **Mic Input Gain -** This is used to adjust the microphone input gain to adjust the drive in SSB mode. It is recommended that the **AUD**io meter be used, setting this parameter for audio peaks above "0dB". *This setting cannot be adjusted if the **MIC** input is not selected.*
- **Line Input Gain -** This is used to adjust the line input gain to adjust the drive in SSB mode. It is recommended that the **AUD**io meter be used, setting this parameter for audio peaks above "0dB". *This setting cannot be adjusted if the **LINE** input is not selected.*
- **ALC Release Time -** This adjusts the release (decay) time of the ALC. A value of 10 is offers modest compression while values of 5 or lower offer fairly aggressive compression. ***See the section about the adjustment of the ALC/Compressor.*** This setting will be displayed in RED and not adjustable unless "TX Audio Compress" is set to "SV".
- **TX PRE ALC Gain -** This is a post-filter, pre-ALC gain setting in the TX audio path

where a setting of 1 is unity. This is increased from unity to increase the amount of ALC action (compression). ***See the section about the adjustment of the ALC/Compressor.*** This setting will be displayed in RED and not adjustable unless “TX Audio Compress” is set to “SV”.

- **TX Audio Compress -** This is the same as the “**CMP**” setting on the main screen and it adjusts the amount of compression of the transmitted audio signal. This parameter dynamically adjusts both “**ALC Release Time**” and “**TX PRE ALC Gain**” to provide a configuration that will result in a small amount of compression for low values or “heavy” compression for high values. When set to “**SV**” *(which would be setting “13”)* the “**ALC Release Time**” and “**TX PRE ALC Gain**” parameters, above, are available for adjustment to provide “custom” processor settings. The “**ALC Release Time**” and “**TX PRE ALC Gain**” settings forced by this parameter are ***not*** saved to EEPROM and the user-configurable settings in “SV” mode are preserved.

CW related items:

REMEMBER: When in the MENU mode, CW timing and speed will be disrupted! Remember this when adjusting parameters such as CW speed and CW TX→RX delay!

- **CW Keyer Mode -** This selects from Iambic-B, Iambic-A and Straight Key modes.
- **CW Keyer Speed -** This allows the adjustment of CW keyer speed, when in Iambic mode, from 5 to 48 words per minute. This is the same as the **WPM** item on the main display screen. ***While you may adjust the CW speed while in menu mode, CW timing and speed will be skewed until you exit menu mode!***
- **CW Sidetone Gain -** This adjusts the sidetone volume in CW mode as well as in the TUNE mode. This is the same as the **STG** item on the main display screen.
- **CW Side/Off Freq -** This adjusts the CW sidetone and TX/RX offset frequency in 10 Hz steps from 400 to 1000 Hz.
    - It should be noted that the CW transmit carrier frequency is always ***higher*** in frequency by ***<u>this</u>*** amount and it ***<u>exactly</u>*** matches the sidetone frequency which is to say that if you match the pitch of the other station's receive signal with the pitch of the sidetone, both with be transmitting on the same frequency.
    - When adjusting the sidetone, always take care to be sure that the center frequency 300 Hz and/or 500 Hz filter that you use matches the sidetone or else the stations that reply to you may do so outside the filter's passband!
- **CW Paddle Reverse -** This swaps the Dit and Dah position of the paddles.
    - Note that if this is turned ON, the “ring” contact of the paddle jack is still the “PTT” line as before.
    - This has no effect when “**CW Keyer Mode**” is set to “Straight Key” mode.
- **CW TX->RX Delay -** This sets the Transmit-to-Receive turnaround time. **Note:** If you experience a problem with the CW key “hanging” occasionally during CW operation *(e.g. it goes “dead” for a second or two and then recovers)* you may wish to increase this time slightly. There ***may*** still be a lingering bug that may show up if the TX->RX turnaround time is set too short, but it is believed that this has been fixed.
- **CW TX/RX Offset** – This sets how the receiver offset and/or the frequency display operates in CW mode according to the following settings:
    - **USB** – The receiver operates in USB and the transmit frequency is ***above*** the displayed frequency by the amount of the configured sidetone frequency *(e.g. menu parameter “**CW Side/Off Freq**”).* One must do some mental math to

calculate the actual transmit frequency.

- **LSB** – The receiver operates in LSB and the transmit frequency is ***below*** the displayed frequency by the amount of the configured sidetone frequency *(e.g. menu parameter “**CW Side/Off Freq**”).* One must do some mental math to calculate the actual transmit frequency.
- **AUT USB/LSB** – In this mode **USB** is selected >= 10 MHz and **LSB** is selected below 10 MHz.
- **USB DISP** – The receiver operates in USB but the displayed frequency shifted ***upwards*** by the amount of the configured sidetone frequency. The displayed frequency is that of the transmit frequency and it is the frequency of the received signal if it is tuned to match the pitch of the sidetone.
- **LSB DISP** – The receiver operates in LSB but the displayed frequency shifted ***downwards*** by the amount of the configured sidetone frequency. The displayed frequency is that of the transmit frequency and it is the frequency of the received signal if it is tuned to match the pitch of the sidetone.
- **AUTO DISP** – In this mode **USB DISP** is selected >= 10 MHz and **LSB DISP** is selected below 10 MHz.
- **USB SHIFT** – The receiver operates in USB. Compared to normal USB for SSB operation, the receive frequency is shifted down and the displayed frequency is shifted up by the amount of the configured sidetone frequency which causes a CW note that would be zero-beat in USB mode to be heard at the pitch of the sidetone frequency. The displayed frequency is that of the transmit frequency and it is the frequency of the received signal if it is tuned to match the pitch of the sidetone.
- **LSB SHIFT** – The receiver operates in LSB. Compared to normal LSB for SSB operation, the receive frequency is shifted up and the displayed frequency is shifted down by the amount of the configured sidetone frequency which causes a CW note that would be zero-beat in LSB mode to be heard at the pitch of the sidetone frequency. The displayed frequency is that of the transmit frequency and it is the frequency of the received signal if it is tuned to match the pitch of the sidetone.
- **AUTO SHIFT** – In this mode **USB SHIFT** is selected >= 10 MHz and **LSB SHIFT** is selected below 10 MHz.

Comments on the various modes:

The “**USB**” and **“LSB**” modes are equivalent to those found on many older transceivers such as the Drake TR-7 in which the transmit frequency was shifted from the receive frequency. In these transceivers the ***actual*** transmit frequency is calculated by adding/subtracting the known frequency offset from the dial frequency.

The “**USB DISP**” and “**LSB DISP**” modes are equivalent to those found on current transceivers such as the Yaesu FT-100, FT-817, FT-847 and FT-897 to name but a few with the “**USB DISP**” being equivalent to the “**CW**” mode and “**LSB DISP**” the same as the “**CW-R**” mode. In these modes the radio's frequency is not shifted, only the display is offset by an amount equivalent to the sidetone frequency. The displayed frequency is the actual carrier frequency of the transmitted signal and that of the received signal if it is tuned so that its pitch matches that of the sidetone.

The “**USB SHIFT**”, “**LSB SHIFT**”, and “**AUTO SHIFT**” operate by shifting both the local oscillator and the display by the amount of the sidetone/offset of the transceiver. Compared to “USB” mode, the display doesn't change at all, but a signal that was zero beat in USB/LSB mode now becomes audible at the sidetone pitch when set to this mode. The “**AUTO SHIFT**” mode is equivalent to the CW mode in many current-production Icom transceivers.

Spectrum Scope related items:

- **Spec. Scope 1/Speed -** This selects the update rate of the spectrum scope, or it may be set to **OFF** which disables the spectrum scope entirely. The **OFF** setting may be used to reduce the “helicopter” sound that may be heard under low-signal conditions. *This has been renamed to “1/Speed” as the lower number indicates a <u>slower</u> speed.*
    - **Note:** The “helicopter” sound may be significantly reduced by placing an **insulated** metal shield between the RF and UI boards.
- **Spec/Wfall Scope Filter -** This adjusts the “smoothing” of the spectrum scope and waterfall display. **Note:** If your board uses an LCD with an SPI interface a smoothing setting of 1 or 2 is recommended.
- **Spec. Trace Colour -** This sets the color of the spectrum trace.
- **Spec. Grid Colour -** This sets the color of the background grid of the spectrum scope.
- **Spec/Wfall Scale Colour -** This sets the color of the frequency scale along the bottom of the spectrum scope and waterfall display.
- **Spec magnify x32 -** When set to **ON** this changes the span of the spectrum scope and waterfall display from its normal +/- 96 kHz to +/- 3kHz. It does not increase the resolution, but rather the thickness of the lines are doubled. Note that in frequency translate mode, the receive (dial) frequency is always placed in the center of the screen.
- **Spec/Wfall AGC Adj. -** This adjusts the AGC response rate of the spectrum scope and waterfall display. The default setting of 10 yields the same response as the previous “fixed” setting of earlier firmware.
- **Spec Ampl. -** This adjusts the number of dB per vertical division that the displayed signal represents. The available settings are:
    - 5dB
    - 7.5dB
    - 10dB
    - 15dB
    - 20dB
    - 1S-Unit (6dB)
    - 2S-Unit (12dB)
    - 3S-Unit (18dB)

  Note that while these settings are primarily for adjusting the vertical scale of the Spectrum Scope, they also have an effect on the brightness and contrast of the waterfall display. With the approximately dynamic range of the visual spectrum display being 4 vertical graticules, a typically useful setting of this parameter is “10dB” as this represents the typical range of signals found on an amateur band under normal conditions.

  It is recommend that you find the optimal setting for the spectrum scope and then <u>leave it there</u> rather than adjust it for the waterfall display, which has its

own set of adjustments for brightness and contrast!

- **Spec/Wfall Ctr. Line** **–** This is used to set the color of the vertical grid line that coincides with the ***center frequency*** of the receiver on the spectrum display and waterfall display to make the "center tuning" frequency more obvious. When **Frequency Translate** is off, this will be in the center, but if **Frequency Translate** is on, this will be to the left and right of the center, depending on whether the mode is set to "**RX LO HIGH**" or "**RX LO LOW**", respectively. If "Magnify" mode is on, this line will always be in the center.
- **Scope/Waterfall –** This parameter has two settings: **SCOPE** and **WFALL** to select Spectrum Scope and Waterfall Display, respectively. There is a "shortcut" to this setting: Pressing the "**BAND-"** and "**BAND+**" buttons simultaneously will toggle between the two modes – although this will have no effect if already in the **MENU** mode.
- **Wfall Colour Scheme** – This selects the color "palette" used to represent the strength of the signals displayed on the waterfall display. At present there are three palettes available:
    - **Grey** – Weak signals are represented by black/very dark colors with strong signals depicted by very light/white colors.
    - **HotCold** – In this palette weak signals are represented by dark blue signals with strong signals indicated by red colors.
    - **Rainbow** – This palette represents weak signals with blue/violet signals with progressively stronger signals indicated as if colors of the rainbow with the red being the strongest.
    - **Blue** – This palette represents weak signal as dark blue, progressing to pale blue as signal strengths increase.
    - **INVGrey** – This is the "inverse" of the "Grey" palette in that weak signals are light and strong signals are dark.
- **Wfall Vert Step Size** – This is the number of vertical pixel steps per waterfall update. While the waterfall data is updated internally each individual pixel, this allows the user to "skip" some internal updates of that data to improve the update rate of the display – particularly if one is using an LCD with an SPI interface. While no visual data is lost by increasing this number, increasing the number too high can cause the display to appear "jerky". A value of "1" is the smoothest as the screen is updated every time new spectral data is available and a value of "2" looks quite smooth.
- **Wfall Brightness** – This adjusts the baseline brightness of the waterfall display. A value of "100" represents zero with numbers above this adding to brightness and those below it subtracting the brightness. If the display is too dark, this value may be increased and vice-versa. This setting is used with "**Wfall Contrast"** to suit the user's taste.
- **Wfall Contrast** – This multiplies the brightness value of the waterfall display where a value of "100" equals 1.00. Increasing this value makes brighter signals brighter and darker signals darker. This setting is used with "**Wfall Brightness"** to suit the user's taste.
- **Wfall 1/Speed** – This adjusts the update rate of the waterfall, with a higher number being a slower rate of update. If you are monitoring a section of an amateur band for activity, you will likely ***not*** want a very fast update rate or else activity on other frequencies may move up the screen too quickly and be missed.

  Note: If the speed is increased *(number decreased)* too much the waterfall speed will increase very little, but the response of the transceiver to button-presses and knob adjustments will become sluggish. Such is indicated by color change of this adjustable parameter from Yellow to Red as this effect will (likely) increase – particularly if DSP is activated.

- **Scope NoSig Adj.** - This adjust how low or high the “no signal” baseline will auto-adjust on the spectrum scope. A low number will raise the baseline up while a high number will lower the baseline.
- **Wfall NoSig Adj.** - This adjust the background and overall brightness of the spectrum scope. A “low” number will brighten the scope while a “high” number will darken the scope. With the waterfall display one may use the “**Wfall Brightness**” and “**Wfall Contrast**” settings to adjust the brightness and contrast of the waterfall display to suit your needs.
- **Wfall Size -** This sets the size of the Waterfall display: **Normal** = The same size as the Spectrum Scope, **Medium** = Slightly larger, without the banner at the top.

Configuration Menu:

The second item of the main menu item is “Configuration Menu”. Open the “Configuration” Menu” using the stylus to access the settings.

General radio setup related items:

- **Rig Model –** Select the transceiver or receiver to be used with the Spectrum.
- **CAT Protocol –** This is set to the default value for the selecte rig.
- **CAT baudrate –** Again set to the default vaue for the rig.
- **CAT mode –** The Spectrum can operate as a Slave, Master or in BiSync mode, where both the transceiver and the Spectrum can control operational frequency and mode.
- **IQ Mode –** The IQ signals can be swapped in software to meet the connection requirements of varios radios. If the IQ signals are reveresed, USB will operate as LSB and visa-versa, and the peaks on the Scope will run in the wrong direction while tuning.
- **CAT TX PTT –** If enabled, PTT commands will be processed via that CAT serial connection, and the EXT DIT/DAH lines will be ignored.
- **Band definition –** The various band definitions for each region can be set.
- **Save out-of-band freq. -** This allows the last frequency to be save during power-down even if it is outside a legal band.
- **TX on out-of-band freq. -** This allows TX activation outside of the legal bands.
- **Transmit disable –** Disable the TX function.
- **Step Size Marker -** When set to ON a line below the appropriate digit of the main frequency display indicates the selected step size.
- **Transmit Disable -** When ON, all transmit functions are disabled. This may also be toggled by pressing-and-holding the **TUNE** button. An indication of **Transmit Disable** being active is the **TUNE** button's text being displayed in grey.
- **O/S Menu SW on TX -** (“On-Screen Menu Switch on Transmit”) When ON several of the receive-specific adjustments (“**AFG**” and “**RIT**”) are switched to transmit-specific adjustments, such as “**CMP**” and **“MIC**” or “**LIN**” in voice modes, respectively. This allows more convenient access to these parameters when in transmit mode. CW-related functions are not available in this manner.
- **Mute Line Out TX** – This enables/disables the muting of the LINE OUT mode when in TX mode. ***The LINE OUT is always disabled when “Frequency Translate” mode is active.***
- **TX Mute Delay –** This causes the transmit audio to be muted for a brief period after activating the PTT line with the settings depicted in 100ths of seconds. The range is from 0 (disabled) to 25 (250 milliseconds.) This may be used to suppress a “click” or “clunk” produced by microphones when the transmitter is keyed, particularly electret types that are powered up at the moment that the radio is keyed.
- **LCD Auto Blank** – With settings of “Off” and adjustable from 5-15 seconds, this enables a feature in which the LCD backlight will automatically blank after the configured number of seconds after a button was pressed or knob turned when ***NOT*** in **MENU** mode. This mode may be used to reduce power consumption – particularly when the transceiver is being battery-powered. When the LCD backlight is turned off, the spectrum scope and waterfall are are also disabled *(“frozen”, actually)*, reducing a potential noise source as well.
- **Filter BW Display –** This setting sets the colour of a line below the Spectrum Scope or

Waterfall display that graphically depicts both the bandwidth and frequency span of the currently selected filter and mode.
- **Voltmeter Cal.** - This setting is used to calibrate the on-screen voltmeter. A setting of 100 (default) represents unity (1.00) with each step representing approximately 0.1%.

Receiver related items:

- **Max Volume -** This sets the maximum permitted setting of the **AFG** ("volume control"), setting the maximum "safe" level. This is most useful to those who exclusively use headphones.
- **Max RX Gain (0=Max) -** This sets the "maximum" gain of the receiver/AGC system. The default of "3" is a compromise of stability in preventing feedback at normal volume levels with no antenna connected. This setting can be used to prevent the receiver's gain from getting too high under no-signal conditions, particularly if all of the various modifications have ***not*** yet been done to prevent feedback. It is recommended that this be set to the default of 3 when Frequency Translate is active.

Beep related items:

- **Key Beep** – When set to **ON** a short beep will be heard with key presses. For "short" (press-and-release) button-pushes, the beep will sound at the instant that the button is released. For "long" (press-and-hold) button-pushes, the beep will sound as soon as the time has elapsed for the press to be considered valid, at which point the button may be released. (This does not disable the "CW Sidetone reference" beep produced by the mode button – *see below*.)
- **Beep Frequency** – This sets the frequency of the key beep, in 25 Hz steps.
- **Beep Volume** – This adjusts the volume (loudness) of ***both*** the key beep ***and*** the beep that is heard when the RIT box is pressed and held (in CW/LSB/USB modes) to generate a CW "reference" tone of the same frequency as the CW transmit-receive offset/CW sidetone.

CAT related items:

- **CAT mode -** This enables the CAT mode which is based on a USB driver that allows remote control of the transceiver. *This setting is* ***<u>NOT</u>*** *saved in EEPROM.* The CAT mode is in development and has limited capabilities.
  - **NOTE:** If you have the USB programming cable connected and enter CAT mode, it is likely that the transceiver will crash! If you have programmed the transceiver you must first disconnect the transceiver for 10-15 seconds before reconnecting it and enabling CAT mode.
  - EEPROM save ***may*** not work reliably if CAT mode has been enabled/disabled since the last power-up.

Frequency related items:

- **Freq. Limit Disable** – This enables the built-in frequency (tuning) limits of 1.8-32 MHz, allowing one to tune the dial/display to practically anything!
  - **NOTICE:** This should be considered to be an ***experimental*** feature and treated

with care, considering the limits of the PLL, the surrounding hardware and the tuning algorithm itself: **There are no guarantees that any of the hardware will work at all outside the "normal" tuning range!**If you enable this feature and tune outside this range, note that the frequency will ***NOT*** be saved with a power-off. Additionally, if you tune outside this range and ***THEN*** turn ***OFF*** this feature, you will find that the band that you were on from which you tuned outside the range ***may be "locked out"*** until you re-enable this feature and tune back down into a "valid" amateur band (e.g. 80-10 meters.) Again, this feature is for ***EXPERIMENTATION ONLY***. If it is reported that it is useful, it and other, related features may be integrated into the transceiver, later.

I/Q Gain and Phase related items:

Notes:

- **Please read the procedure for calibration of the RX IQ gain and phase balance for more detailed information.** This procedure may be found elsewhere in this document.
- You must be in the appropriate mode (e.g. LSB, USB, RX, TX) in order to adjust the relevant item. If the item is available to be adjusted, its parameter will be displayed in white.

- **LSB RX IQ Bal.** - This adjusts the IQ Gain balance in LSB RX mode.
- **LSB RX IQ Phase.** - This adjusts the IQ Phase balance in LSB RX mode.
- **USB/CW RX IQ Bal.** - This adjusts the IQ Gain balance in USB/CW RX mode.
- **USB RX IQ Phase.** - This adjusts the IQ Phase balance in USB RX mode.
- **AM RX IQ Bal.** - This adjust the IQ Gain balance in AM RX mode.
- **LSB TX IQ Bal.** - This adjusts the IQ Gain balance in LSB TX mode.
- **LSB TX IQ Phase.** - This adjusts the IQ Phase balance in LSB TX mode.
- **USB/CW TX IQ Bal.** - This adjusts the IQ Gain balance in USB/CW TX mode.
- **USB TX IQ Phase.** - This adjusts the IQ Phase balance in USB RTX mode.

Transverter related items:

- **XVTR Offs/Mult -** This is a transverter multiplication factor that can range from **OFF** to **1-10**. When this parameter is set to something other than OFF, the multiplication factor and the offset (below) is applied and the digits of the main frequency are displayed in **YELLOW.**

  Note: When the transverter mode is active, 1 MHz and 10 MHz frequency step sizes will be available for tuning the main frequency dial.

- **XVTR Offset (Hz) -** This is the frequency offset that is applied to the transverter multiplication factor. A frequency offset of up to 999.000 MHz may be "dialed" in.
  - When this parameter is selected within the menu system it is possible to use the **STEP** buttons to select a step size of 1 and 10 MHz. When one navigates away from this parameter in the menu and a 1 or 10 MHz step size is selected, a smaller step size will automatically be forced.

The above offsets the display as follows:

DSP related items:

- **DSP NR BufLen** – This is the length of the De-Correlation delay buffer. In order for the DSP to tell a voice signal from noise, it must have a sample of each, but given the absence of a separate noise source, we have to “simulate” one by delaying the original signal to “de-correlate” it. If we delay it too little, it will resemble the voice too much and be ineffective. If we increase the delay, we can improve the performance but if we delay too much we end up with an “echo” type effect and a sluggish response.
    - **This value must always be <u>larger</u> than “DSP NR FFT NumTaps”, below. If this rule is violated, the number will turn RED and DSP NR operation will become ineffective.**
- **DSP NR FFT NumTaps** – This is the number of taps in the FIR (filter) comprising the DSP noise reduction filter. A smaller number of taps implies a more agile filter, but also one that is less accurate while a larger number of taps is more precise and potentially slower to respond: A more “precise” filter may also reduce the actual performance in that the automatic calculation of the filter's parameters – which are, by their nature, imprecise, may “miss the mark”. ***<u>A higher number will increase processor loading and slow the user-interface response.</u>***
    - **This value must always be lower than “DSP NR BufLen”, above. If this rule is violated, the number will turn RED and DSP NR operation will become ineffective.**
- **DSP NR Post-AGC** – This determines whether the DSP noise reduction will take place ***before*** the audio filtering and AGC or ***after*** the audio filtering and AGC. The net effect will be the same, but there will be important differences as perceived by the user:
    - **“NO”: DSP Noise reduction takes place *before* filtering/AGC** – The operation of the DSP noise reduction **will** affect the S-meter reading. Because the noise reduction occurs prior to the AGC, the “quieting” caused by the noise reduction will be compensated by the AGC and the perceived “quieting” effect caused by the noise reduction will be reduced. **Note that this can give the *<u>impression</u>* that the noise reduction is less effective than it actually is!**
    - **“YES”: DSP Noise reduction takes place *after* filtering/AGC** – This operation of the DSP noise reduction ***does not*** affect the S-meter reading. If very “heavy” noise reduction is occurring, this can cause the perceived audio level to drop, requiring that one “rides” the volume control, particularly if there are weaker signals, buried in the noise, amongst strong – a situation that can exaggerate the volume differences! **Be careful if you are wearing headphones when using this setting!**

  Comment: It is recommended that one make judicious use of the “RF Gain” control (**RFG**) to reduce the receiver gain when using the DSP – particularly if “**DSP NR Post-AGC**” is set to “**NO**” - to reduce the amount of noise that is heard under “no signal” conditions.
- **DSP Notch ConvRate** – This adjusts the convergence factor (“mu”) of the filter and will have an effect on how quickly it “attacks” a CW note. Because of the nature of the filter, this parameter's effects aren't as obvious as those of the “Strength” adjustment of the noise reduction filter. The higher the number, the more quickly it will “attack” and notch a tone that appears in the passband. It should be noted that very high numbers

(e.g. a configuration to "attack" a tone very quickly) can also affect voice quality.

- **DSP Notch BufLen** – This is the length of the De-Correlation delay buffer. In order for the DSP to tell a CW note from noise, it must have a sample of each, but given the absence of a pristine noise source, we can "simulate" one by delaying the original signal to "de-correlate" it. If we delay it too little, it will resemble the original signal too much and be ineffective and start to affect voice. If it is increased, the notch becomes more accurate, but it can slow down and, for a number of reasons, actually lose effectiveness.

Noise Blanker related items:

- **NB AGC T/C (<=Slow) –** This is the time constant for the noise blanker AGC and it may be adjusted in an effort to improve the performance of the AGC. A lower value corresponds with a slower AGC within the noise blanker algorithm.

IMPORTANT: While you are in the menu system and it is being displayed, the noise blanker is ***always*** disabled, so ***you must exit the menu to note the effect of that parameter!***

Transmit audio related items: (set-up dependent)

- **AM TX Audio Filter** – When set to "**ON**" this will insert a "brick wall" audio bandpass filter (275-2700 Hz, approx) in the audio transmit path – the same filter that is used on SSB transmit. When set to "**OFF**" this filter is bypassed, allowing audio to be passed below 100 Hz and somewhat above 3000 Hz, improving fidelity.
    - Note that while fidelity is improved when this filter is turned off, "talk power" is reduced as more transmit energy is devoted the parts of the voice that do not carry information.
- **SSB TX Audio Filter** – When set to "**ON**" this will insert a "brick wall" audio bandpass filter (275-2700 Hz, approx) in the audio transmit path – the same filter that is used on SSB transmit. When set to "**OFF**" this filter is bypassed, allowing audio to be passed below 100 Hz and somewhat above 3000 Hz, improving fidelity.
    - Note that while fidelity is improved when this filter is turned off, "talk power" is reduced as more transmit energy is devoted the parts of the voice that do not carry information.
    - If the SSB audio filter is disabled, ***<u>PLEASE be considerate</u>*** of other users on the amateur bands as your signal will also be "wider", extending both above 2.7 kHz and also suffering somewhat in opposite-sideband rejection to approximately 200 kHz.

FFT Spectrum Scope and Waterfall related items:

- **FFT Windowing** – The use of "FFT Windowing" - the pre-processing of spectral data before display – can greatly improve the visual performance of the Spectrum Scope and Waterfall display by decreasing "bin leakage" (e.g. "sidelobe") - that is, the tendency of a signal (say, a carrier) to “leak” onto the display above and below the displayed frequency.

The use of “FFT Windowing” - the pre-processing of spectral data before display – can greatly improve the visual performance of the Spectrum Scope and Waterfall display by decreasing “bin leakage” (e.g. “sidelobe”) - that is, the tendency of a signal (say, a carrier) to “leak” onto the display above and below the displayed frequency.

The default setting is “Blackman” which is quite good in making the spectrum scope look sharper and preventing signals from “bleeding” into each other, but most of these windowing functions take a degree of processor time and can slightly slow the response to knobs and buttons, particularly if the waterfall speed is increased to “Yellow” or “Red” speeds.

The following windowing functions are available:

- **Rectangular:** This is the same as ***no*** window function and results in rather poor visual performance, particularly in the presence of strong signals amongst weak ones. This setting has negligible processor load.
- **Sine (a.k.a. “Cosine Window”):** This is slightly narrower than “Rectangular” but still fairly wide and causes minimal processor loading.
- **Bartlett (a.k.a. Fejér):** A “Triangular” window – somewhat better than “Rectangular” and minimal processor load.
- **Welch (Parabolic):** Comparable to Bartlett.
- **Hann (Raised Cosine – fastest as enhanced)**: Very good sidelobe rejection, but not as narrow as Hamming or Blackman. This has higher processor load than the previous functions.
- **Hamming (Raised Cosine):** Narrower than Hann, but not quite as good sidelobe rejection.
- **Blackman (Default):** This is a good, general-purpose function with about the same “narrowness” as Hamming but not quite as good sidelobe rejection.
- **Nuttall:** Slightly wider than Blackman, comparable sidelobe rejection. This has the highest processor loading of the available functions.

It may be observed that with the “wide” window function (Rectangular, Sine, Bartlett, Welch – in decreasing tendency) that strong signals will tend to “smear” across the display.

In contrast, for the narrowest function (e.g. Hamming and Blackman) these can cause some weaker signals to be a bit more difficult to see as with wider, more spread-out energy of voice signals may not be integrated into multiple FFT bins and may not be quite as visible.

The function with the best sidelobe rejection is the Hann, but even though it is wider than the Hamming or Blackman this can also cause some weaker signals to become more difficult to see owing to its excellent sidelobe rejection and the lack of strong spectral energy from one particular bin leaking into adjacent bins and being “visually integrated”.

In experimentation I have found that the Blackman is the most visually appealing, offering a reasonable compromise between sidelobe rejection, width, and overall "look".

For more information than you ever wanted to know about FFT Windowing, see the article:

http://en.wikipedia.org/wiki/Window_function

[End of menu configuration items]

Notes about adjustment of DSP-related configuration values:

DSP Noise Reduction parameters:

The DSP Noise Reduction is active in either the DSP **NR** or **NR+NOT** mode and it performs noise reduction be detecting the coherent (e.g. non-random) properties of the human voice and quickly adapting a filter to pass those frequencies and blocking the other frequencies.

The "strength" of this filter may be adjusted using the menu item #10, "**DSP NR Strength"** - but be very careful with this as it easy to go overboard with this setting. If it is set too high, the artifacts caused by the noise reduction *(e.g. a "hollow" or "watery" sound)* can be ***worse*** than the interference than you are trying to remove!

The default setting is a good place to start, and carefully increase experimentally on signals of varying quality to get a "feel" the effects.

It should be noted that "**DSP NR BufLen"** and "**DSP NR FFT NumTaps"** will also interact with the efficacy of the **"DSP NR Strength"** setting, sometimes making a particular "strength" setting weaker, sometimes making it "stronger.

Again: Remember that the "DEFLT" button will restore the settings to usable defaults!

DSP Automatic Notch Filter parameters:

The DSP Notch filter is an "Automatic" notch filter that will immediately "seek and destroy" any CW (continuous) carrier that it finds, but it should have a minimal effect on the normal human voice. It is active in the "**NOTCH**" and "**NR+NOT**" modes, but it is <u>***always disabled***</u> when in the CW mode as it would make such operation impossible.

The notch filter operates within the signal path ***prior*** to the AGC and the **DSP NR** operation, so a strong "tune up" signal will not cause the S-meter to deflect when the notch filter is active, but note that the AGC is still active and the receiver may still de-sense if this signal is very strong and cause the lower half of the S-meter to flash red.

Also note that the presence of a strong carrier may also cause some "intermodulation" distortion – both from mixing products within the transceiver's analog circuitry, but also due to the dynamic limitations of the A/D converter as well as artifacts in the mathematical calculations being carried out in the SDR itself!

Note:

- The notch filter may be useful in AM mode to eliminate the "tweet" that appears when tuned very close to the center frequency. If you are listening to a shortwave broadcast station, note that the automatic notch may occasionally "attack" music with interesting results!

Operation at very high NR "strength" settings (e.g. >= 35):

As the DSP “strength” setting is increased the rate of filter adaptation is slowed down. While this can have the effect of make a filter “stronger” to a degree by making it focus more strongly on the voice components rather than the rapidly-changing noise, if this setting is increased too much it may change too slowly to track ***different*** voices!

While the higher settings (e.g. >= 35 or so) may (or may not) be useful for voice, they can be useful for narrowband signals that do not exhibit fast changes, spectrally speaking – such as CW: The effects of very “strong” DSP settings on CW signals can, under certain circumstances, be quite striking!

With very high “strength” settings and the slow adaptation rate, one may perceive that the filter may be “stuck”, but turning the DSP filter off and then back on will “reset” it and cause it to re-train. If you are using the the DSP NR filter at such high settings, it is worth experimenting with turning it off and on to get a “feel” as to how the filters respond.

It should be noted that at very high DSP settings (>45) the DSP NR is more susceptible to crashing when exposed to strong impulse noises: Refer the the section about automatic and manual resetting of the DSP NR, below. At these high settings the DSP may “crash” by producing a loud white noise rather than go completely silent.

Important Note:

> If, when the a menu item is changed, it will be necessary to press the MENU then PWR button to save the changes to the EEPROM.

Approximate specifications of the Spectrum DSP:

The following specifications are for the Spectrum M2 version 1.0 with firmware 1.00e.

Because this <u>is</u> a software-defined device and due to ongoing modifications/improvements of the software and hardware, the specifications continue to improve!

- **IQ sensitivity for 10dB S/N, CCITT filtering, taken at 28.3 MHz:**
  - **Frequency Translation enabled:** Better than -111 dBm (0.6uV) in a 2.3 kHz bandwidth, better than -126 dBm (0.11uV) in a 300 Hz bandwidth
  - **Frequency Translation Disabled:** Better than -108 dBm (0.89uV) in a 2.3 kHz bandwidth, better than -120 dBm (0.22uV) in a 300 Hz bandwidth

    The above specifications are for a receiver on which the published sensitivity modifications are performed.
- **Frequency coverage:** Receiver/Transceiver dependent
- **Spectral Display Modes:**
  - **Spectrum Scope:** This is a spectrum analyzer with the vertical divisions representing user-definable amplitude variations of 5, 7.5, 10, 15, 20, 1 S-Unit (6dB), 2 S-Units (12dB) or 3 S-Units). The baseline *("reference level")* of the analyzer is automatically adjusted so that the signals within the displayed passband best-fit the dynamic range selected by the user-selected dB/division. A graticule along the bottom of the display indicates the approximate frequency of the signal being displayed over a width of +/- 96 kHz. (192 kHz total.)
  - **Waterfall Display:** As with the spectrum scope, the frequency is displayed along the "x" axis but the signal strength is implied by the displayed color. The newest signals are displayed along the bottom of the screen, but as new readings arrive, the representations of the older signals are shifted upwards giving an ephemeral time record of recent activity on nearby frequencies. There are several options for color "palettes" that range from simple grayscale to "cold-hot" to "rainbow" to represent weak to strong signals.
  - There is also a "Magnify" or Zoom mode for both the Spectrum Scope and Waterfall Display mode that provides x32 magnification, reducing the visible spectral width from 192KHz to just +/- 3 kHz (6 kHz total).
  - Both the Spectrum Scope and Waterfall Display are highly configurable. It is possible to disable one or both spectral display modes if desired.
- **Large-signal handling capability:** Continuous "Clip Warning" occurs above approximately -28 dBm and ***actual*** A/D clipping and distortion occurs at and above approximately -18 dBm for signals +/- the local oscillator frequency and higher for signals outside this range.

- **Available TX/RX modes in this firmware version:** CW, USB, LSB, AM *(full-carrier, double-sideband)* and FM. *AM transmit and FM transmit/receive capabilities are available ONLY if the "frequency translate" is activated (highly recommended!)*
  - **FM options:** Carrier (ultrasonic) squelch, subaudible tone encoding and decoding, tone burst ("whistle up") generation, "narrow" (+/-2.5 kHz) and "wide" (+/-5 kHz) deviation and the selection of 7.2, 10, 12 or 15 kHz pre-detection receive bandwidths.
  - **FM sensitivity for 12 dB SINAD, CCITT filtering:**
    - **7.2 kHz BW filter:** -103.7 dBm (1.46uV) with 1 kHz tone at +/-1.5 kHz
    - **10 kHz BW filter:** -102.1 dBm (1.75uV) with 1 kHz tone at +/-1.5 kHz
    - **10 kHz BW filter:** -104.0 dBm (1.41uV) with 1 kHz tone at +/- 3kHz
    - **12 kHz BW filter:** -102.7 dBm (1.63uV) with 1 kHz tone at +/- 3kHz
    - **15 kHz BW filter:** -99dBm dBm (2.50uV) with 1 kHz tone at +/- 3 kHz
- **CW mode receive/transmit and frequency display details:** Nine modes of CW display/shifting are available to emulate the various makes of radios and suit the user's taste, ranging from no shifting, display-only shifting, display and LO shifting and manual or automatic LSB/USB shifting.
  - In CW mode "**CW-L**" or **"CW-U**", is displayed depending on whether LSB or USB is being used for reception.
- **CW Speed range:** 5-48 WPM.
- **Available audio filter bandwidths in this firmware version:** 300 Hz, 500 Hz, 1.8 kHz, 2.3 kHz, 3.6 kHz, with a "wide" filter of 5, 6, 7.5 ***or*** 10 kHz being selectable in all modes except FM, where the filtering is done prior to demodulation as noted above. *All filters are software-defined and additional bandwidths could be made available.*
- **DSP Filtering Capability:** Noise reduction and Automatic Notch Filter with adjustable parameters. *Notch filtering is disabled in CW mode for or when using a "wide" receive bandwidth.*
- **S-Meter calibration:** "Industry Standard" *(IARU Region 1, Technical recommendation R.1)* S-meter calibration where S-9 = -73dBm (50.2uV @ 50 ohms) with each "S" unit representing 6 dB. Units above S-9 are in dB units, as noted. For more information on this calibration system see the article: **http://en.wikipedia.org/wiki/S_meter**
- **External audio output connections:** Headphone ports are 3.5mm 3-pin Jack connectors to allow the connection to an external device. With these connectors it is possible to interface with an external device *(a computer or tablet/smart phone)* and operate "Sound Card" modes with the Spectrum DSP such as SSTV, PSK31, WSPR and other analog/digital modes.
- **CAT RS232 levels:** 3.3V TTL. If RS232 are required, as for Elecraft KX3, an RS232 connector with converter is available.
- **EXT input DIT/DAH signal levels:** 3.3V TTL.
- **Transmit ALC type:** Look-ahead gain compressor with both pre-set and available "custom" settings.

- **Operating voltage range:**
  - 4 - 5.0 volts maximum
  - As low as 3.5 volts, receive-only.

- **Current consumption:**
  - **Receive:**
  - Approx. 140-160mA at 4.40 volts, minimum volume, maximum display brightness.
  - The selection of minimum LCD display brightness can reduce this by 40-60mA.
  - The modification to the PA drivers to switch off bias when not in TX mode can reduce this by a further 30-60 mA.
  - **Power off:** <50 mA standby mode.

The ALC (Automatic Level Control)

(Set-up dependent)

Prior to the addition of the ALC the **POWER** adjustment on the Spectrum DSP was somewhat irrelevant when in a voice mode as it only added effective attenuation in the audio path. If one adjusted the audio to 5 watts PEP when in the 5 watt mode, it was possible to switch to the 1 watt mode and readjust the audio gain to again achieve 5 watts as there was nothing within the code to set levels!

What is more significant is that there was nothing in the code to prevent the overdriving of the final amplifier stage, even if it had been set up properly for a “clean” 5 watts as there was no way to be sure, without using an external RF power meter, that the transmitter audio drive was properly set.

This was been changed in version code 0.0.207: **It is no longer possible to obtain a higher PEP power at a given power setting than a steady carrier in CW or TUNE mode! Unless you have a true peak-reading RF power meter, you *will* read a lower RF output power in SSB mode than in CW mode.**

Please re-read the above paragraph at least once to be sure that you understand it!

How the ALC works:

All modern SSB transceivers have a form of ALC which monitors the transmit power level and if it exceeds the set power level, it is cut back to prevent overdriving of the finals. In this way the ***maximum*** output power may be set for a mode that has intrinsically varying power levels.

With the ALC the PEP power from the transmitter should not exceed the carrier level observed in TUNE mode, no matter the audio drive level.

In order for the ALC to work there must be at least a ***minimum*** audio level to drive it and to provide for this clicking the **SWR** meter will change to one of three modes:

- The **SWR** meter. This dynamically measures the forward and reflected RF power, calculates the VSWR and displays it.
- The **AUD**io meter. This shows the audio level from -20dB to +12dB, with 0 dB being “nominal”. It is acceptable for audio to occasionally peak at +6 to +10dB.
- The **ALC** meter. This shows the amount of ALC action, from 0 to 34 dB – *more on this below.*

Adjusting for the proper audio level when in SSB transmit mode:

(Setup dependent)

- Speak normally if using the Microphone input, or set the nominal input level if you are using the LINE Input mode.
- Use button **F2** to select the **AUD**io meter.
- Use button **M1** (below **ENC1**, the left-hand encoder) to select the on screen **CMP** setting and use that encoder to adjust it to a setting of 1.
- Use button **M3** (below **ENC3**, the right-hand encoder) to select the on-screen **MIC** (or **LIN**) setting and use that encoder to adjust it, or you may go into the Menu mode and adjust the "**Mic Input Gain**" (or "Line Input Gain" as appropriate).
- While speaking normally, adjust the gain so that the audio meter peaks up to "0" (zero) to +6 on the audio meter. It is fine for it to occasionally peak higher than this.
- Now use button **F2** to select the **ALC** meter.
- Use button **M1** (below **ENC1**, the left-hand encoder) to select the on screen **CMP** setting and use that encoder to adjust it, or you may go into the Menu mode and adjust the "**TX Compress Level**".
- Adjust this setting for an upwards indication of the **ALC** indicator. ***See below for a discussion of this setting.***

Using the ALC to control transmit power, or as a speech processor:

There are two ways to adjust the speech processor/compressor settings:

- Using the "**CMP**" ***numerical*** settings *(which is the same as the "**TX Audio Compress**" menu parameter)*
- Setting the "**CMP**" *(or the "**TX Audio Compress**")* setting to "**SV**" and independently adjusting the "**ALC Release Time**" and "**TX PRE ALC Gain**" settings.

Using numerical settings for CMP:

When using the numerical settings for the **CMP** setting *(also the "**TX Audio Compress**" parameter)* the "**ALC Release Time**" and "**TX PRE ALC Gain**" settings are automatically adjusted to provide "compression" settings that become "stronger" with an increasing number.

"Manually" adjusting parameters when "CMP" is set to "SV":

When the "**CMP**" *(or the "**TX Audio Compress**" parameter)* are set to "**SV**" the parameters "**ALC Release Time**" and "**TX PRE ALC Gain**" may be manually adjusted as desired to provide a custom compressor setting.

This ALC system has been designed to be flexible and be usable both as a "standard" ALC used to set the SSB transmit power ***and*** as a highly-effective compressor-type speech processor. To operate the the ALC in this way requires attention to two separate parameters as described below.

SSB operation with minimal speech compression:

- Set the Microphone/Line gain as described in the previous section *(e.g. around "0" on the **AUD**io meter with occasional peaks to +6 to +10.)*
- In the menu system, set the parameter **ALC Release Time** to the default setting of **10**.
- While speaking normally adjust the **TX PRE ALC Gain** parameter for a peak reading on the **ALC** meter of 4-6 dB.
- Setting **ALC Release Time** to a higher value will reduce the compression even more.

SSB operation with maximum speech compression:

- Set the Microphone/Line gain as described in the previous section *(e.g. around "0" on the **AUDIO** meter with occasional peaks to +6 to +10.)*
- In the menu system, set the parameter **ALC Release Time** to the default setting of **3 or lower**.
- While speaking normally, adjust the **TX PRE ALC Gain** parameter for a peak reading on the **ALC** meter of 8-16 dB.
- Setting **ALC Release Time** to a lower value and the **TX PRE ALC Gain** to a higher value will increase the compression even more.

Explanation of the parameters and meters:

- **Mic Input Gain/Line Input Gain:** These operate directly on the microphone and line inputs in the way that you would expect. These parameters display as **MIC** or **LIN** on the main display, respectively.
- **Audio meter:** This displays the audio level, in deciBels, on the selected audio input, with "0" being the level that will ***just*** achieve 100% power at the bottom of the ALC threshold. The level displayed is ***NOT*** filtered in any way and signals outside the frequency range that would be transmitted (e.g. <200 Hz, >3500 Hz) will register.
- **TX PRE ALC Gain:** This is a variable audio gain ***after*** audio filtering in the transmit bandpass, ***after*** the audio metering, above, but ***before*** the ALC circuit.
- **ALC Meter:** This indicates the amount of gain ***reduction*** in deciBels that the ALC is providing to the audio path. The ALC is in the audio path ***after*** transmit audio filtering so it will not respond to audio that is outside the frequency range that will be transmitted. The ALC can only ***reduce*** gain *(by up to 40 dB)* but it can <u>never</u> increase it and it will settle to unity under no-signal conditions. *Note that the ALC meter's response **<u>is</u>** indicative of the ALC release time – see below.*

- **ALC Release Time:** This sets the time, after audio has dropped below the current threshold, that the ALC will take to release and reduce attenuation. When set to the default setting of 10, the ALC will have only a modest effect on the transmitted audio, taking several seconds for the ALC to completely recover from a voice peak while setting it to the maximum value if 20, the effect is almost that of disabling the ALC entirely in terms of added compression in that the gain recovery rate is approximately 1dB/second. Low values (below 5) will “follow” audio very quickly and offer effectively very high compression rate.

Warnings: (set-up dependent)

- Do not set the Mic/Line gain such that the peak audio level on the **AUDIO** meter regularly peaks much above 4 to 8B, although occasional peaks to +10 are fine. Avoid settings that “peg” the meter as this could result clipping and audio distortion.
    - If the RF amplifier is working properly and not being overdriven, input audio clipping should not cause “splattering” on the transmitted signal, only “bad” sounding audio.

REMEMBER:

If your RF power meter does not have a "Peak" reading function specifically designed to read PEP on SSB signals *(many do not!)* it will always give a false "low" power reading on SSB, which is to say that your power on voice ***peaks*** may be where it should be, but your meter will be reading a much lower pseudo-average!

Amplitude Modulation (AM) mode reception and transmission:

Receive “Frequency Translation” or offset mathematically shifts the center frequency by +12 kHz, depending on the setting of menu item “**RX/TX Freq Xlate”**. This function solves the problem of the “Zero Hertz” hole that would otherwise cause a nulling of the AM carrier if the receiver were tuned such that it were placed in the center of the receiver's passband. ***For more information on this problem, see the information later in this section.***

If “**RX/TX Freq Xlate”** is enabled, there are no special considerations that need to be taken when tuning an AM signal other than those noted in the section below regarding the narrow AM filtering.

Note: When offset-tuning and AM signal using the wide bandwidth filter mode with frequency translate mode active, it is possible, when off-tuning by 12 kHz to place the AM carrier in the “Zero Hertz” hole, resulting in distortion of the received signal.

Tuning AM signals with wide and narrow filtering:

The AM bandwidth filtering operates as follows. ***The bandwidths noted below are <u>always</u> available in AM, regardless of menu settings***:

There is one “wide” bandwidth and several different bandwidths are available for this setting:

- **10 kHz:** Pre-detection bandwidth: +/-10kHz (20 kHz total); Post-detection bandwidth: 10 kHz.
- **7.5 kHz:** Pre-detection bandwidth: +/-7.5kHz (15 kHz total); Post-detection bandwidth: 10 kHz.
- **6 kHz:** Pre-detection bandwidth: +/-6kHz (12 kHz total); Post-detection bandwidth: 10 kHz.
- **5 kHz:** Pre-detection bandwidth: +/-5kHz (10 kHz total); Post-detection bandwidth: 10 kHz.

Other (“non-wide”) bandwidths are available:

- **3.6 kHz:** Pre-detection bandwidth: +/-3.6 kHz (7.2 kHz total); Post-detection bandwidth: 3.6 kHz.
- **2.3 kHz:** Pre-detection bandwidth: +/-2.0 kHz (4.0 kHz total); Post-detection bandwidth: 2.3 kHz (300-2600Hz, adjustable).
- **1.8 kHz:** Pre-detection bandwidth: +/-2.0 kHz (4.0 kHz total); Post-detection bandwidth: 1.8 kHz (500-2300Hz, adjustable).

Some explanation is required for the **1.8 kHz** and **2.3 kHz** modes as you'll note that the pre-detection bandwidth appears to be a bit on the narrow side to accommodate the sidebands that extend out beyond the filter (e.g. greater than the +/-2kHz bandwidth). If one tunes the receiver to the center frequency of the AM signal when these bandwidths are enabled the audio response will be limited to just 2 kHz by the pre-detection filter. If one off-tunes from the center frequency, this +/- 2 kHz bandwidth – which encompasses 4 kHz - may be shifted to include the higher audio frequencies of one or the other sidebands of the AM signal.

Because it is always necessary to off-center tune an AM signal to obtain the full audio bandwidth permitted by the **1.8 kHz** or **2.3 kHz** post-detection filter, one of the two sidebands (upper or lower) may be encompassed in the narrower bandwidth. This “quirk” may also be used to advantage in the presence of QRM (interference) by selectively tuning for one sideband or the other, moving away from the source of the interference.

In this version the filtering when in AM mode has been re-done: The Hilbert transformers, which have a bandpass response, are replaced with low-pass filters *(e.g. response down to DC)* that have their low-pass cut-off frequency selected according to the desired bandwidth. Post-detection, there is additional audio filtering applied to reduce the wideband noise that inevitably results with envelope detection of weak signals.

The “Zero-Hertz” hole problem if operating with “Frequency Translation” disabled:

This (and all “sound-card”) type SDRs have a “hole” at zero Hertz – right in the middle of the display. This is the inevitable result of AC coupling to the A/D converter (codec) and cannot easily be helped without added design complication.

What this means is that if you tune in an AM signal “dead center” its carrier will fall into this “hole” and disappear which effectively turns it into a ***double sideband with no carrier*** – which is to say, it is ***no longer AM!*** **If an AM signal is tuned dead-center, it will sound distorted – much like an SSB signal tuned on an AM receiver!**

The solution is simple: **Do NOT tune the AM signal so that the carrier is "dead center"**. It is necessary only to offset-tune by a few hundred Hertz, but ***it is*** necessary to do this!

Comments on adjusting the AM RX IQ Balance:

This adjusts the receiver I/Q amplitude balance when in AM mode and is used to minimize the low-level "tweet" *(e.g. tone)* that may be heard when an AM signal is tuned in slightly off center frequency. To null this tone it is recommended that you tune in a strong carrier, offset it by 500 Hz and then adjust this parameter to minimize the amplitude of this tone.

This adjustment is unlikely to completely eliminate this "tweet", but it can significantly reduce it. Note also that the efficacy of this reduction changes with audio frequency in that the optimal null for a 400 Hz "tweet" tone *(e.g. 200 Hz offset from the carrier frequency)* will be different from that of a 1000 Hz "tweet" tone.

AM Transmission: (installation dependent)

AM transmission is possible ONLY when frequency translation mode is active!

When transmitting using AM, the power level will automatically be set to 2 watts to prevent the PEP from exceeding the maximum "clean" power level available. *If one does not transmit, the power level will not be automatically changed.*

If an attempt is made to transmit with frequency translation mode turned off, the transmitter will key, but there will no transmit out power at all in AM mode!

There is no "TUNE" mode in AM mode!

You should remember several things about AM:

- It is **MUCH** less efficient than SSB! You will have a 9 dB reduction (1/8th) of the "talk" power of SSB: That's just the way it is!
- The **UNMODULATED** resting carrier will be **25%** of that of the **peak** power! This means that if you are used to getting 5 watts peak on SSB, you will get only 1.25 watts when no audio is present: Sorry about that, but that's just the laws of physics!

The speech processor works in AM mode in exactly the same way that it does in SSB mode and it should ***NOT*** be possible to exceed 100% modulation.

There is presently ONE option for AM transmit mode: In the configuration menu, the item labeled "**AM TX Audio Filter**" has the selection of **ON** and **OFF**. If it is "**ON"** *(default)* the transmit audio will be "brick-wall" filtered from about 275 to 2700 Hz in the same way that the SSB audio is.

If this selection is set to "**OFF**" the audio filter is disabled. This has the effect of increasing the fidelity of the audio – mostly through additional low-frequency components (down below 100 Hz) and somewhat above 3000 Hz. While this can increase the audio fidelity on transmit, you should be aware that it can significantly shift the RF energy from the audio spectrum that contains speech intelligence and reduce the "talk power".

Frequency Modulation (FM) mode reception and transmission:

IMPORTANT:

- **The FM mode is installation dependent and is disabled by default**.
- **The noise blanker and DSP noise reduction and notch are disabled in FM mode**

WARNING:

- **Transmitting on FM means that a continuous carrier is being generated. Be absolutely certain that the final transistors on the Spectrum DSP are adequately heat-sinked and do not get too hot and also that your power supply is capable of the current being drawn.**
- **It is recommended that the *lowest* transmit power be used on FM that will achieve a "full quieting" signal for the receiving station's (or repeater's) receiver.**

FM (Frequency) modulation is now available ***experimentally*** on the Spectrum DSP and while it is believed to work properly, it likely has a few "issues."

Repeater operation is possible if one uses the "SPLIT" mode and sets the receive and transmit frequencies in separate VFOs. By "swapping" the VFOs, one can effect a "reverse" function and listen on the input frequency while transmitting on the output – useful for checking for a simplex path.

The "FM" produced by the Spectrum DSP is compatible with PM, which is to say that for modulation, a 6dB/octave pre-emphasis is applied and a 6dB/octave de-emphasis is done on receive audio: **This is the worldwide standard for narrowband frequency modulation on amateur and commercial frequencies.**

To enable FM:

- You **MUST** have Frequency Translate enabled: It will not work without it!
- Set the menu item **"FM Mode Enable"** to **ON.**

When in FM mode, the "**RFG**" control becomes "**SQL**" (Squelch) – also operated by middle band swiping. The higher the number, the "tighter" the squelch and a setting of "0" unconditionally opens the squelch. You will note that when the squelch opens, the "FM" on the mode indicator will change color: *More on this later.*

Two "modes" of FM are available: The default is "Narrow" with +/- 2.5 kHz peak deviation (w/1kHz modulation) and ""Wide" with +/- 5 kHz peak deviation, the former being commonly used on HF and the latter being that which is used on VHF bands except for those instances where the "narrow" (+/- 2.5 kHz) mode is specifically used by local option.

These two modes are more or less interoperable, with the following provisions:

- Operation of "Wide" (+/-5kHz) on "Narrow" frequencies will result in "loud" audio, possible "squelch clamping" and splattering onto adjacent "narrow-spaced" channels.
- Operation of "Narrow" (+/-2.5kHz) on "Wide" frequencies will result in chronically "low" audio, reduced "copyability" under weak-signal (noisy) conditions and everyone telling you to talk louder!

There is also a selection of receiver bandwidths:

- **7.2 kHz** – This is suitable only for “Narrow” operation and even this will result in a bit of added distortion as the filter about as narrow as it may be to pass audio. Using this setting the weak-signal sensitivity is the highest amongst the FM filter bandwidth settings since this narrowest filter also intercepts less noise under weak-signal conditions.
- **10 kHz** – This is the “default” bandwidth and is suitable for “Narrow” bandwidth and while it will work for “Wide” bandwidth, slight distortion may occur on voice peaks.
- **12 kHz** – This is wider than necessary for “narrow” bandwidth and is recommended for “Wide” bandwidth.

Subaudible tone generation:

The menu item “**FM Sub Tone Gen**” enables and sets the frequency for subaudible tone generation. All common frequencies (including NATO and others) are included and when set to something other than OFF, the tone is modulated on the carrier during transmit.

The tone deviation in “Narrow” mode is approximately +/- 300 Hz and around +/- 600 Hz in “Wide” mode.

Subaudible tone detection:

Also known as “Tone Squelch” this is enabled by setting the menu item “**FM Sub Tone Det**” to something other than “Off”. When it is activated, it is required that **BOTH** the squelch be open AND the tone be detected.

Comment:

> Sub-audible tone detection, particularly in the presence of noise, is rather difficult. If the tone detection drops in and out on weak signals you may wish to turn it off.

Tone Burst generation:

Although it is getting rarer, some repeaters may still require “tone burst” and this is enabled by setting the menu item “**FM Tone Burst**” to something other than “Off”. Two tone frequencies are currently provided: 1750 Hz and 2135 Hz. The duration of the tone is 1 second.

To transmit a tone burst:

- Key the transmitter. (It must be in FM mode)
- While the transmitter is keyed ***<u>Press-and-Hold</u>*** the Mode button (set-up dependent).
- While a tone burst is being generated the **“FM”** on the mode indicator changes to a yellow background.

Squelch/Tone decode indications:

The “FM” mode indicator changes according to the squelch and tone detection status:

- **“FM” on dark blue background** – Squelch closed (and tone decoder not decoding).
- **“FM” on light background** – Squelch open.
  - If tone decoding is ***not*** active, audio will be heard.
  - If tone decode is active, audio will ***not*** be heard, but this indicates that a signal is present (or the squelch is loose) but does not match the tone decode setting.
- **“FM” on red background** – This is displayed if the tone decoder is detecting a tone: Audio will be heard.

Additional comments:

- The audio (speech) processor is active on FM, but owing to the nature of FM communications (e.g. low noise with good signals) “strong” compressor settings will likely result in audio that sounds vary “processed.” Lower settings (e.g. “2”) are recommended when using FM.

- The squelch range is a bit “compressed” - that is, a there is not much difference in the low number range for squelch adjustment (for weak/noisy signals) but only a small range for those that are nearly full quieting. It is possible that at the maximum squelch setting that it will "clamp" on audio peaks or may even fail to open at all, in which case one should reduce the setting.

- The tone decoding is a bit slow to respond, particularly on the “release” when a tone disappears – in which case a properly-set squelch will close the audio gate. This is an artifact of the need for narrow bandwidth detection, the need to validate the tone detection to prevent it from “bouncing” and also the limited amount of processing power available to do the tone decoder processing.

Initial set-up of the Spectrum DSP:

There are a number of set-ups that can be optionally performed/checked, including:

- **Carefully observe the start-up screen:** If you see warnings – typically in red – pay close attention to and resolve them – some of which are detailed in this section.
- **Turn on the "Frequency Translate" mode [TUNE/O shows green] for best receive quality if noise interference is detected.** .
- **Frequency Calibration:** (for Spectrum with the RF/IF receiver module installed)
    - A known-accurate frequency source. This could be a precision oscillator/test equipment or an off-air signal from a time station such as WWV.
    - A means of accurately determining an audio frequency. This could be a frequency counter or a computer with sound card with a known-accurate sample rate running an audio analysis program.
  - Set the receiver to USB mode.
  - If the signal source is available on multiple frequencies, pick the ***highest*** frequency available.
  - Tune the receiver 1 kHz ***below*** the frequency of the selected reference frequency.
  - Set the tuning step size to 1 Hz.
  - Couple the output of the receiver to the audio frequency measuring device so that you get a reading.
  - Go to the calibration menu item "**Freq. Calibrate**" and adjust it to obtain a precise 1 kHz audio frequency.
    - **Note:** It is possible that the Si570 may be off frequency by hundreds of Hz – or even several kHz. ***You may change the frequency step size*** to facilitate this adjustment.
- **TX and RX I/Q Phase/Balance adjustments:**
  - The procedures for adjusting TX and RX I/Q Phase and balance are detailed in later sections of this manual and should be reviewed.

Additional settings:

The following settings should be known to the Spectrum DSP operator. Please refer to the previous sections containing the menu items for more details on their use.

- **Voltmeter Cal.** - This calibrates the on-screen voltmeter: Only a known-accurate voltmeter should be used as a reference. Note that the voltmeter display updates only once every second or so.
- **Max Volume** – This puts a limit in the maximum volume setting – particularly useful if one primarily uses headphones where loud sounds may cause hearing damage.

Recommended procedure for adjusting RX IQ gain and phase adjustments – *Frequency translate mode EN<u>ABLED</u>:*

1. Set the Spectrum DSP to LSB mode.
2. Set the AGC to FAST mode so that the receiver recovers more quickly from the "clicks" that occur during the phase adjustments.
3. Tune in a strong, constant signal. This could be a shortwave broadcast station or a signal generator – a signal generator is preferred.
4. Tune the Spectrum DSP dial frequency 1 kHz above the carrier frequency to obtain a strong 1kHz audio note. *Example: If your signal generator is set to 7200 kHz, you would tune to 7201 kHz.*
5. Now tune the dial 2 kHz lower, that is to a frequency 1 kHz below that of the signal generator: You may hear a 1 kHz tone "leaking" through from the other side. *Following the example, the dial frequency would now read 7199 kHz.*
6. If you hear a tone, adjust "**LSB RX IQ Bal.**" to minimize the amount of leakage. Now adjust "**LSB RX IQ Phase**" to further minimize the signal leakage. If no obvious improvement is obtained, set this to the default value (0). In any case, note the two values obtained.
7. If you have selected the "**RX LO LOW**" mode of frequency translation, tune the dial 11 kHz ***higher*** than the generator's frequency. *Following the example, the dial frequency should now be 7211 kHz.* If the "**RX LO HIGH**" mode of frequency translation is selected, tune the dial 13 kHz ***lower*** than the generator's frequency. *Following the example, the dial frequency would now be 7187 kHz.*
8. If you hear the "leakage", adjust "**LSB RX IQ Bal.**" and "**LSB RX IQ Phase**" alternatively to minimize it. Once minimized, note the two values obtained.
9. Use the set of values obtained during the adjustment that caused the ***most*** difference when adjusting both the **"LSB RX IQ Bal.**" and "**LSB RX IQ Phase**" settings, with preference on the values obtained in **step 7**.
10. Repeat the steps, above, in USB mode, adjusting the parameter "**USB RX IQ Bal.**" and "**USB RX IQ Phase**".
    - For this procedure the frequency used in **Step 5** would modified such that one would tune 2 kHz above the frequency where a 1 kHz tone would be heard, or 1 kHz above the signal generator frequency. *(The dial frequency would be 7201 kHz, using the example above.)*
    - If "**RX LO LOW**" translate mode is used, the frequency used in **Step 7** would be 13 kHz *above* that of the frequency generator. *(7213 kHz, using the example above.)*
    - If "**RX LO HIGH**" translate mode is used, the frequency used in **Step 7** would be 11 kHz *below* that of the frequency generator. *(7189 kHz, using the example above.)*
11. Repeat the steps, above, in AM and FM modes, adjusting the parameters "**AM RX IQ Bal.**" and "**FM RX IQ Bal.**", respectively.

Once you have completed the procedure, remember to set the AGC mode back to what it had been previously and to save the new settings by pressing-and-holding the POWER button to turn off the transceiver.

Recommended procedure for adjusting RX IQ gain and phase adjustments – *Frequency translate mode DISABLED:*

Do not perform the following procedure unless you operate the Spectrum DSP with "Frequency Translate" set to "disabled" for a specific technical reason. Because using "Frequency Translate" improves most aspects of operation of the transceiver it is expected that it will normally be used. This procedure is included for completeness.

1) Set the Spectrum DSP to LSB mode
2) Set the AGC to FAST mode so that the receiver recovers more quickly from the "clicks" that occur during the phase adjustments.
3) Tune in a strong, constant signal. This could be a shortwave broadcast station or a signal generator – a signal generator is preferred.
4) Tune the Spectrum DSP dial frequency 1 kHz above the carrier frequency to obtain a strong 1 kHz audio note.
5) Now tune the Spectrum DSP dial frequency 2 kHz lower (e.g. 1kHz below) the carrier frequency. You should be able to hear the same 1 kHz audio note, but much more weakly.
6) If you can ***NOT*** hear this note, re-check the frequency. If the frequency is correct and you cannot hear the "leakage", either the test signal is not strong/clear enough or your opposite sideband attenuation is sufficient and you should proceed to adjusting the USB gain/phase adjustments.
7) If you hear the "leakage", adjust the **LSB RX IQ Bal.** to minimize it.
8) Once minimized using the **RX IQ Bal.**, adjust the **RX IQ Phase** to further minimize the "leakage". Note that adjusting the phase will cause "clicking" which may upset the AGC/S-meter briefly.
9) **Once the LSB leakage has been minimized, repeat the above procedure in USB mode, but tuning below in step 4 and above in step 5.**

Once you have completed the procedure, remember to set the AGC mode back to what it had been previously and to save the new settings by pressing-and-holding the POWER button to turn off the transceiver.

NOTE for CW operators who use "lower" CW sidetone frequencies:

If you use the Spectrum DSP primarily for CW, use rather low frequency CW notes and sidetone frequencies *(400-550 Hz)* and notice "leakage" from the opposite sideband after following the above procedure, you may choose to perform the above procedure at the approximate frequency CW sidetone frequency rather than 1000 Hz. This is because of the way the Hibert Transformer works and the fact that lower frequencies (<500 Hz) can have poorer opposite-sideband rejection.

If you choose a different, lower side-tone frequency note that you may sacrifice opposite sideband rejection at higher frequencies, particularly if you null it at too-low a frequency! You should carefully choose your "alternate" frequency as to provide a good compromise good opposite sideband rejection at the desired frequency and higher frequencies (e.g. 750 Hz and up).

Explanation of the "Frequency Translation" feature

PLEASE read the following VERY carefully!

Menu item "**RX/TX Freq Xlate**" selects the enabling/disabling of baseband frequency translation in the receiver/transmitter. When the translation is active, instead of the receiver operating at and around "DC", the signals are mathematically shifted from 12 kHz (above or below – user-selectable). When frequency offset/translate is active, the **[TUNE/O]** button will be shown in green.

Performing this frequency shift can help forgive a lot of the "sins" that occur with "DC" conversions - the most obvious of which are that ANY noises in the power supply as well as the 1/F noises of op amps, mixers, A/D converters and the like tend to show right up in the received audio. With the signals at microvolt levels, minimizing these signals requires careful design and earthing considertions!

- These signals/problems can show up as:
  - Hum
  - Buzzing with the dimming of the backlight
  - Noises from the Waterfall SPI communications (e.g. "ticking")

It should be noted that these code modifications ***DO NOT*** relieve the builder of the ***strong recommendation*** that one perform the modifications in the "Spectrum DSP Board Modifications" file, particularly the U3a and MCU and LCD power supply modifications (for UI board 0.3) but they should go a long way toward reducing the artifacts that can still occur even after making those modifications - even to the point of gaining an extra S-unit or two in sensitivity.

Menu item "**RX/TX Freq Xlate**" has the following options:

- **OFF** - This is the original operation of the transceiver with the receive (and transmit) signals operating at and around zero Hz.
- **RX LO HIGH** - In this mode the signals are shifted BELOW zero Hz by 12 kHz, requiring that the local oscillator be shifted up by the same amount. The received signals are tuned at the first graticule left of center on the spectrum scope and waterfall display when "magnify" mode is turned off.
- **RX LO LOW** - In this mode the signals are shifted ABOVE zero Hz by 12 kHz, requiring that the local oscillator be shifted down by the same amount. The received signals are tuned at the first graticule right of center on the spectrum scope and waterfall display when "magnify" mode is turned off.

Additional options may be added in the future.

The use of **RX LO HIGH** or **RX LO LOW** is entirely up to the operator's preference, but it may be observed that additional rejection of some of the noises *(e.g. the “tick” from the SPI communications of the waterfall update to the LCD)* may be reduced when one setting is selected over the other.

For various reasons (e.g. the use of USB on higher bands where the potential for zero-HZ interference is highest) the use of “RX LO LOW” is recommended for best performance!

Frequency translation mode is <u>necessary</u> for AM TX and for FM RX/TX operation.

Side-effects of the frequency translation mode:

Spectrum Scope/Waterfall Display offset:

If the menu item “**Spec/Waterfall magnify**” is set to **Off** when the translate mode is activated, you will note that the receive signal is ***no longer in the center of the spectrum scope or waterfall display,*** it is shown at it's offset frequency. Along the bottom of the spectrum scope you'll observe that the frequency display is changed, with the frequency in kHz being displayed in full under the graticule, being shifted left or right as noted above. ***If you have used other SDR software – particularly “sound card” SDR rigs on computers – you will already be familiar with this sort of shift!***

If “Magnify” mode is activated, the receive frequency is always displayed in the center of the screen.

Slip-tune mode: [Slip/SP] quick press & the button shows green when active.

In this mode the stylus is used to drag the receiving frequency and it's carrier bar, across the spectrum, which if not zoomed, is 192 Khz wide. You can instantly tune across the band without changing the transceiver frequency, when [Slip/SP] is pressed again, and the CAT connection is in use, the transceiver will be returned to the current frequency ready for use.

Translation in transmit mode: (for Spectrum's fitted with TX module)

This frequency translation is used on SSB transmit as well, slightly improving the SSB audio quality when this mode is activated and it also makes it possible to implement the transmission of AM signals using this type of hardware.

Effects in AM mode:

The use of frequency translation [TUNE/O long-press] removes the DC Zero offset interference that may occur when using “AM” mode, eliminating the need to off-tune the signal to prevent its carrier from being at “zero IF”. As noted above it has also allowed the implementation of full-carrier, double-sideband AM: It is also possible to implement single-sideband, full-carrier AM, but this feature is not implemented at this time.

Spectrum DSP Key function matrix

Button(s)	Brief press	Press-and-hold >=1 second
MENU/PWR	Power off. Click confirm.	No function set.
Left zone	Select **AFG** and **CMP** or **STG**	Click DSP, click zone box for mode & swipe in regions.
Middle zone	Select **RFG** and **DSP** or **NB**	Switch between **DSP** and **NB**
Right zone	Select **RIT** and **MIC** or **LIN**	Switch between **MIC** or **LIN**
Bandscope hdr	Left-side of scope hdr	Toggle Spectrum Scope and Waterfall modes

Main operational (receive/transmit) mode:

F1	Enter MENU mode	Save settings to memory
F2	Enter Snap mode	Balance the spectrum display to RX (ant. disconnected)
F3	Enter Slip tune mode	Split working mode
F4	Toggle VFO A/B	Activate fast virtual dial tuning mode
F5	Toggle TUNE mode	Toggle SDT zero offset frequency (+12KHz/0)

Menu mode:

F1	Exit MENU mode	Save settings to memory
F2	Set selected menu item to default	No function set.
F3	Move to previous menu screen	Move to beginning of current menu
F4	Move to next menu screen	Move to end of current menu
F5	PWR (click to shutdown)	No function set.

- Pressing the LCD screen while powering up the radio will cause defaults to be loaded. The user must either disconnect power to retain the old settings ***or*** continue to overwrite them with defaults. ***All configurations, adjustments, frequencies and mode settings will be reset to default!***
- Pressing the LCD immediately after the small blue square appears on boot, will enter the LCD calibration mode. Follow the instructions to complete, or disconnect the power to retain.

Operational notes, and known issues:

- DSP Noise reduction enabled: When both scope and Trace are enabled, the speed of the screen updates falls due to the extra processing requirements. If DSP NR is enabled the situation may start to overload the MCU, the Trace will then be automatically disabled to reduce the load, if the load continues to be excessive, Noise Reduction will be disabled. We are developing new firmware to improve the efficiency of the trace feature and noise reduction which will be available as an update.
- When the LCD display is dimmed, it is a known that the display may appear to flicker. This is due to the PWM control method employed.
- Strong signal breakthrough. Depending on the filtering in the transceiver or receiver, alias signals can be seen and heard. The G90 in particular is known to suffer from stations break-through and an external bandpass filter can resolve the issue.
- Xiegu IQ output: This is a very low-level signal and requires the external IQ amplifier to work. The Xiegu pre-amplifier should be enabled to improve signal detection and demodulation. If very strong signals are present, disabled the preamplifier if station breakthrough occurs.
-

Circuit description of the Spectrum DSP:

The Spectrum DSP consists of one main unit and several optional additions: The main unit contains the MCU (processor), memory, audio codec, various sensors and a separate LCD touch display unit.

IQ amplifier: Provided as an external cable, providing 28dB of gain for low-level IQ output of Xiegu, Elecraft and other radio equipment.

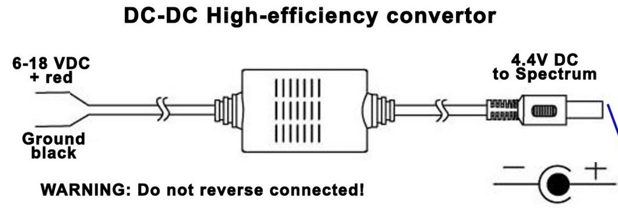
DC-DC convertor: Provides a high efficiency conversion of 8-18V DC to 5V DC as required by the Spectrum and external audio amplifier.

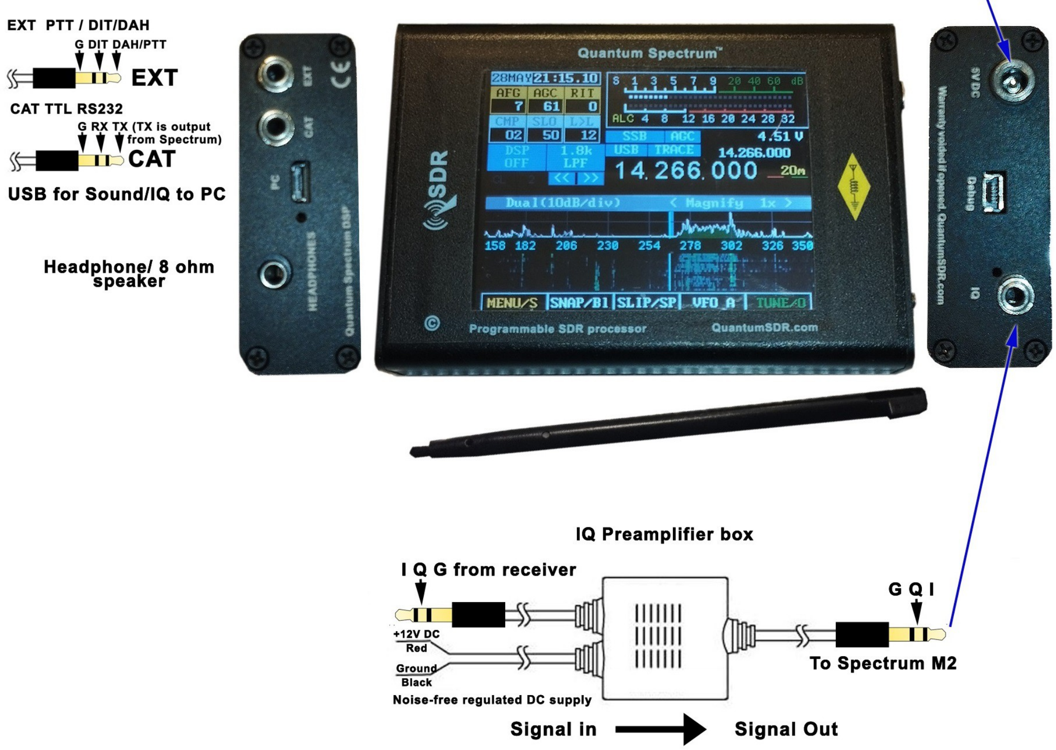
Speaker amplifier: Provided as an external cable to provide a louder audio output to larger speakers.

Spectrum DSP M2 Connection Instructions

Requirements:

The CAT cable **must be connected** for the Spectrum to **show the correct frequency and mode**, and for commands to reach the transceiver. The **headphones and speaker** output must not be grounded.





Instructions: - PLEASE TAKE CARE INSERTING CONNECTORS TO AVOID BREAKAGE!

1. Without plugging the power cable into the Spectrum DSP, connect the DC-DC convertor power input cable to a regulated 8 to 18 V DC source. A 200MA constant current is required. Check that the power output on the Output lead reads 4.4V DC. Do not plug it in yet. You can also connect a regulated 5V DC supply directly, but don't connect yet.

2. Connect the preamplifier to your radio's IQ port or tap. Then connect the preamplifier output jack to the IQ port on the Spectrum. You can also connect an IQ signal directly to the Spectrum DSP providing it has around 30mW PP for weak stations. USDX & (tr)usdx do not require a preamplifier, but the Xiegu G90 for example, does.

3. Power on your radio set, then plug the DC-DC output cable to the Spectrum, you should see the QSDR logo appear followed by the spectrum display, where, all being correct, you will see signals. At QuantumSDR.com you can find information and videos on using the M2.
If you have any questions please email us: info@quantumSDR.com

XIEGU NOTE: Power-up your Xiegu at the same time as the Spectrum, else it may detect any CAT communications as firmware reloading. Otherwise disconnect the CAT cable during boot.

Supplemental information

The Spectrum M2 DSP is the fruit of a project aimed at bringing affordable Digital Signal Processing and Spectrum displays to radio enthusiasts, allowing classical and modern radios to be updated to a full range of modern features.

In addition, the project, under the "Quantum SDR" banner, aims to provide voice interactive radio control for senior enthusiasts whose sight isn't what it used to be.

Clearly, an undertaking of this nature is technically challenging and with limited resources the deliveries may take many weeks or months in some cases until the production line has got things streamlined!

NOISE INTERFERENCE

Nearly all Software-defined radio (SDR) systems are using an offset from the LO frequency, from what is called DC-zero, where the technology is susceptible to locally-generated low-frequency noise.

With a good power supply and grounding system, the Spectrum DSP can operate without the offset, but this requires careful attention to grounding, and in some cases keeping the Spectrum chassis separate from the station ground.

Station breakthrough: Strong broadcast stations can cause interference in SDR. Use a band-stop filter if needed, or modify the filter in the preamplifier, which may require a capacitor change.
Xiegu G90's preamp can be prone to strong station breathrough. Options: Disable it or use a band-stop filter. We are working on filters add-ons to improve this issue.

How to use the Spectrum:

It's up to you, plug it in with or without the CAT connection and it can be used as the Spectrum display of your radio, nothing more needed to do!

But for many radios sets, the M2 offers better audio quality, with much less noise, plus DSP noise reduction and an excellent notch filter implemented by the Universal Ham SDR team, so that's one reason to use the Spectrum interface.

Another reason is the Slip-tuning mode, where you can use the Spectrum touch display to rapidly tune across nearly 200 KHz of band, and when you find a station you want to talk to, press a button and your transceiver is instantly synchronised; this also eliminates wear on your rigs encoder.
One benefit of Slip-tune mode is that it uses an offset frequency, so local noise is automatically reduced or usually totally disappears.

XIEGU G90 WARNING: Always power-up the Xiegu before the Spectrum if CAT used, as at boot the Xiegu can enter firmware update mode if it sees data on the port.

Thank you for your patience and support!

We are working to improve and add new features to the Spectrum every month.
Please don't hesitate to contact us by email if you have suggestions.

For support please contact info@quantumSDR.com

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