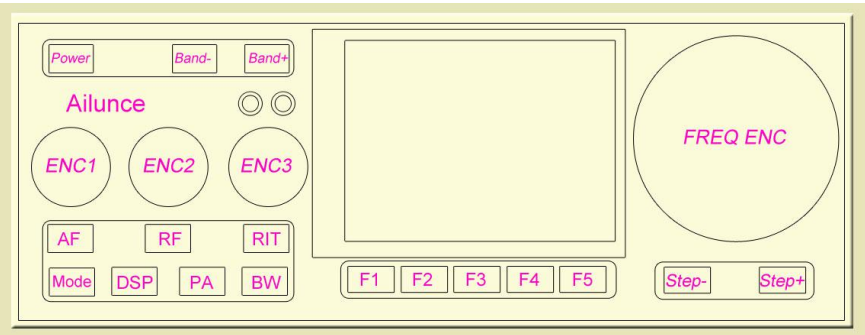


Ailunce **HS1** USER'S MANUAL



Preface:

This manual is for the HS1 transceiver, a good color tft display screen for it and good trx. As an open-source SDR (Software Defined Radio), both in terms of Software and Hardware. As such, the features of this transceiver will continue to evolve and this manual is intended to provide a reference source.



Parameter:

Frequency Range	RX: 0.5MHz-30MHz TX: 0.5MHz-30MHz
Output Power	Maximum 15W
Receiving Sensitivity	Better than -115dBm
Operating Mode	Full Modes
Operating Voltage Range	DC 9-15V
Dimensions	190mm*69mm*45mm
Maximum Current	3.5A
Transceiver comes with automatic key generator circuit, Can be connected to the use of automatic keys.	

Main Features

- Spectrum Dynamic Waterfall Display
- Multiple Working Modes: Receive Mode, Transmit Mode, TUNE Mode, VFO Mode, SPLIT Mode
- DSP Digital Signal Processing Noise Reduction
- Automatic Notch Filter
- Receive Fine Tuning Function, Changeable MIC Gain Value
- VCC Power Supply Voltage Indication Table
- Transmit Signal Strength Display Table
- Multifunctional Instrument: SWR Standing Wave Ratio Meter, AVD Audio Frequency Meter, ALC Signal Modulation Meter

Front Panel controls:



Figure 1: Front panel controls of the HS1 transceiver

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

Power ~ This turns the transceiver on, but it is also used to turn the transceiver off *and save configuration and frequency mode/memories*. A brief press of this button will also select the brightness of the LCD backlight. *Please read notes about the backlight and the possibility of its injection of a tone into the receiver when a “dim” mode is selected.* Low brightness will cause the noise inside the machine to increase, and there is no such noise at 100% brightness.

Important:

If you have not already done so, please find the menu item “RX/TX Freq Xlate” and set it to “RX Low LOW” as this will improve receive and transmit performance. *See this item's menu description in this manual for more details.*

Band-, Band+ ~ These buttons select the next lower or higher amateur band. When the lowest or highest band is reached it “wraps around” to the highest/lowest band. Pressing and holding the Band- along with the Power button may be used to turn on/off the automatic backlight blanking feature while pressing and holding the Band- and Band+ buttons will toggle between the display of the Spectrum Scope and Waterfall Display

STEP-, STEP+ – This sets the tuning step size in steps that include 1 Hz, 10 Hz, 100 Hz, 1 kHz, 10 kHz, and 100 kHz. The function of these buttons may be swapped via a menu setting. Pressing and holding of one of these buttons will temporarily change the step size to facilitate tuning in smaller or larger steps while pressing-and-holding both of these buttons simultaneously will toggle “frequency lock” on and off, with “on” being indicated by the main Frequency readout being displayed in grey.

FREQ ENC – This is used to tune the transceiver's operating frequency, the tuning steps being set by the STEP- and STEP+ buttons.

ENC1, AF – Rotary encoder **ENC1** is typically used to adjust the volume, but its function may be changed using button **AF** to adjust the side tone gain

ENC2, RF – Rotary encoder **ENC2** is typically used to adjust the RF gain (or squelch in FM mode but its function may be changed using button **RF** to adjust the action of the DSP Noise Reduction or Noise Blanker strength. In the Menu mode it is

used to select the item to be adjusted. Pressing-and-holding button **RF** when in normal (non-menu) receive mode will switch between the right-hand function adjusting the DSP Noise Reduction or the Noise Blanker “strength.”

ENC3, RIT – Rotary encoder **ENC3** is typically used as an **RIT** (Receiver Incremental Tuning) but its function may be changed using button **RIT** 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. In the Menu mode it is used to modify the item selected, or button **RIT** may be pressed-and-held to select whether Microphone-In or Line-Input mode is active and to be adjusted.

Mode – This button is used to select the operating mode of the transceiver (CW, USB, LSB, etc.) Pressing this button cycles through the available modes. Pressing-and-holding this button will allow the selection of a mode that is disabled in the menu system (e.g. AM.)

When “LSB/USB Auto Select” is enabled, pressing button Mode 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 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 button Mode - AM will then be selected.

DSP – This button is used to control the DSP audio filter mode. Pressing-and-holding will turn DSP on/off while preserving the current settings. Pushing this button will also “reset” the DSP.

PA – This button is used to set the transmit power level (FULL, 5 Watts, 2 Watts, 1 Watt, 0.5 watts, and back to FULL.)

***Note:** The power is automatically limited to 2 watts in AM Transmit mode. When in CW, LSB or USB mode (those modes with a “BFO”) pressing-and-holding this button will cause a tone with a frequency equal to that of the CW sidetone and transmit shift to be sounded.*

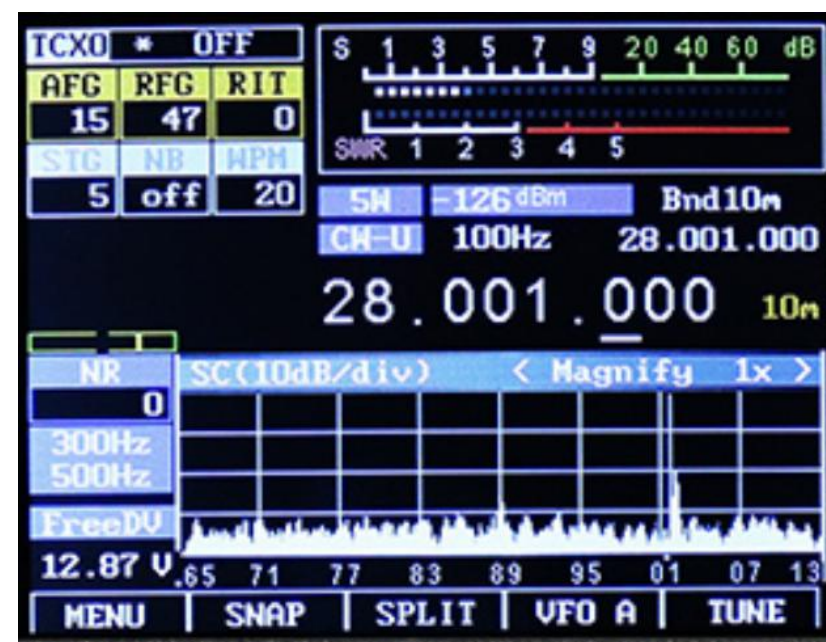


Figure 2: Main display of the HS1 transceiver

BW – This button is used to select the audio pass band filter of the receiver. Pressing-and-holding this button will force the selection of bandwidth that are otherwise disabled. The filter-selection function of this button is disabled when the 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.

Buttons **F1-F5** are “soft” buttons located under the display, the functions of which change depending on mode, indicated on the LCD itself and will be discussed in more detail later in this document.

Also on the front panel are two LEDs, **LD1** on the left and **LD2** on the right. **LD1**, which is typically green, is illuminated on receive and **LD2** which is typically red is illuminated on transmit.

Main Display:

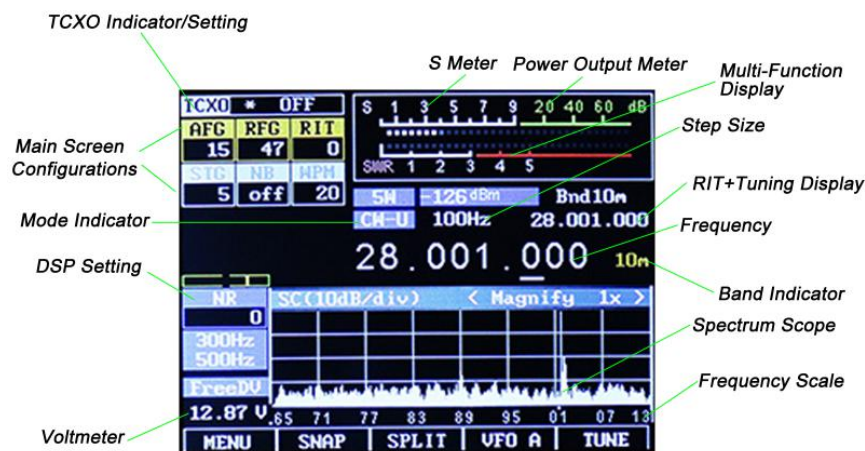


Figure 3: The main screen (annotated) with the SPLIT function activated.

When “Frequency Translate” mode is on, the center frequency indicator will be shifted to the left or right of center by 6 kHz.

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 2**) or as a “Split” display as shown in **Figure 3** with separate transmit and receive frequencies. If the numbers in this display are grey the “Frequency Lock”(toggled by pressing-and-holding both the **STEP-** and **STEP+** buttons simultaneously, or 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 display is an indicator of the amateur band in which the current frequency is tuned. If the current frequency is outside an amateur band it will display “Gen” (e.g. “General Coverage”).

- **Mode Indicator:** Above the “10's” digit of the main frequency display is the current mode displayed on a blue background.

- **Step Size Indicator:** Above the center of the main frequency display, between the Mode Indicator and the “Sub” Frequency display is the setting of the current step size. In Figures 2 and 3 the step size is set to 1 kHz. Optionally, a “marker” may be activated that puts a line under the digit indicating the currently-selected step size (see the menu item “Step Size Marker”).

Along the top there are a number of additional indicators:

- **TCXO Mode/Display:** In the top-left corner the “TCXO” box indicates whether the TCXO (Temperature-Compensated Xtal Oscillator) is active or not. The TCXO is used to read the temperature of the Si570 synthesizer (U8 on the RF board) - which should be

thermally-bonded to the temperature sensor, U10, with a piece of copper or aluminium - and apply a compensation to it to keep on frequency. When it is active the bar graph below the temperature display will display white dots with a blue marker that moves about but when set to “Off”, the bar graph will be grayed out. If set to “Stop” the temperature display will be replaced with “STOPPED”. In Figures 2 and 3 the TCXO is set to ON and displaying a temperature of 112.5F, but this may be set to display the temperature in Centigrade. If the temperature is very low (below 0C or 32F) this will display dashes and the temperature compensation will be disabled until the temperature-coupled synthesizer/sensor exceed this minimum threshold.

- **S-Meter:** This S-meter is nominally calibrated so that S-9 equals 50 micro volts 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, using button F2, 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.

Along the left-hand edge there are a few more indicators, starting from the bottom-left corner:

• **VCC:** Below this is a voltmeter that indicates the current supply voltage. Below 9.50 volts, the digits are displayed in red, orange below 10.5 volts and yellow below 11.0 volts. The receiver may function as low as 8 volts, but obtaining more than 3-5 watts of “clean” transmitter power may not be possible below 10.5 volts, particularly on the higher bands.

• **FIL:** Below this is the current filter bandwidth setting, selectable by using button BW. In Figures 2 and 3 the bandwidth is shown being set to 2.3 kHz.

• **Power Output Setting:** Just above the FIL icon is the currently-selected output power setting, selectable using button G3. In Figures 2 and 3 the power is shown being set to 5 watts.

• **DSP Setting:** Just above the Power Output Setting is the indicator of the DSP mode. The modes available are: “OFF”, “NR”(Noise Reduction), “NOTCH”, and “NR+NOT”(Noise Reduction and Notch).

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.

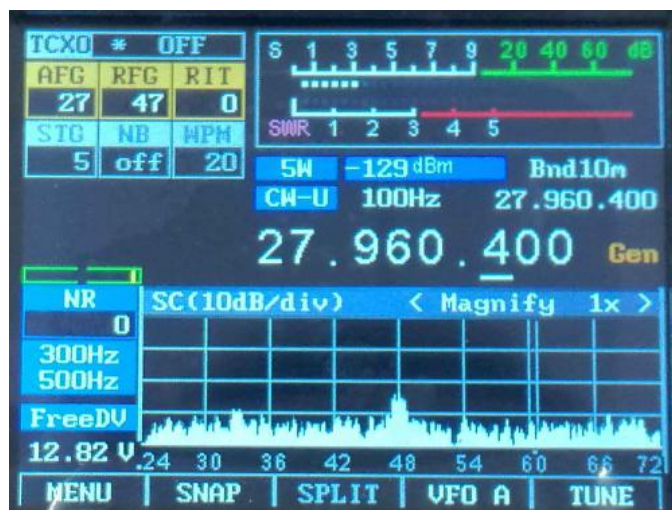


Figure 4: A typical waterfall display in the "Magnify" mode showing +/- 12 kHz (24 kHz) of a band segment. When "magnify" mode is **NOT** active +/-24 kHz (48 kHz) of a band is visible.

Options available to both the Spectrum Scope and 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 +/- 24 kHz or +/- 12 kHz, with the scope's AGC operating only on signals within the displayed span – see the menu setting “Spec. 2x 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:

- Line Out (J1). This is a receive audio output that is fixed level (unaffected by the volume control) that may be used to supply audio to a computer for “sound card” (digital) modes. This connector also carries the audio being transmitted.
- Line In (J2). This input may come from a computer for “sound card” (digital) for transmitting. Its use must be selected in the menu for it to be used.

- Microphone/PTT (J3). This connector has both a microphone connector with bias voltage (if R68 is installed) for powering an electret microphone and a PTT (Push-to-Talk) line that is shorted to ground to key the radio. While the PTT line is always active, the microphone must be selected as the active audio input from the menu for it to be used. (Note that this is also the “Dah” line, which is also used for keying in CW “straight key” mode.)
- Speaker/Phones (J4). This jack feeds and external speaker/headphones, disconnecting the internal speaker when something is plugged into it.
 - Warning: There is no limiting resistor in series with this audio connection, so you must remember to turn down the volume before plugging in headphones.

On the RF board, along the right side, there are three connectors. Starting from the top these connectors are:

- Power connector (J1). This is a coaxial power connector, 5.5mm O.D., 2.1mm I.D., that supplies power to the transceiver. The outer shell is negative and the inner conductor is positive.
- Paddle (J2). This connects to either a set of Morse paddles or a straight key. The outer conductor (“ring”) is typically the “Dah” while the tip is the “Dit” when in Iambic mode. In “Straight Key” mode only the outer conductor (“ring”) is used. (The “Dah” line is the same as the “PTT” line.)

- Accessory (J3). This is used for interfacing with an external device and may be used for keying the transmitter and/or determining when the transmitter is keyed. The “tip” of this jack is the “PTT”/“Dah” line and may be used when interfacing the transceiver to a computer when operating a digital mode. The outer conductor (“ring”) is grounded when the transceiver is in transmit mode and this may be used to key an external amplifier or TR switch.

On the left-hand side of the UI board are two USB connectors.

- The upper, “A” type (full-sized) USB connector is a USB host port that may have future use for storage of data/audio files and/or interface devices such as keyboards and wireless devices. It is also possible to load firmware from a USB memory stick from the interface.
- The lower “mini” USB host port is primarily used for programming firmware into the transceiver.

Finally, the sole connector on the left-hand side of the RF board is the BNC-type antenna connector, the nominal impedance being 50 ohms.

Operational modes and functions:

Receive mode:

After powering up, the HS1 transceiver 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**. In this mode **LD1**, the left-hand LED (typically green) is illuminated.

By default, ENC1 controls the volume, ENC2 the RF Gain and ENC3 controls the RIT.

Transmit mode:

When in transmit mode LD2 (typically red) the right-hand LED is illuminated. In transmit mode most of the controls are frozen, this being done to prevent the change of frequency, filter type and mode during mid-transmission.

TUNE mode:

Tune mode may be entered by pressing the button located below the TUNE icon on the screen (e.g. button F5) at any time and in this mode a carrier is generated, along with an audible sidetone in the speaker, the amplitude being set by the “Sidetone Gain” (STG) setting. The output power may adjusted during transmit by pressing the button PA to cycle through the settings. The TUNE label on the LCD will turn red while TUNE mode is active.

Always have a suitable load connected to the transmitter (matched antenna or dummy load) before entering TUNE mode or ANY transmit mode.

Pressing the TUNE button again will exit.

Notes:

- When in TUNE mode audio being input to the Microphone and LINE inputs will be ignored.
- When TUNE is activated in SSB mode, the frequency offset from the display frequency and the sidetone frequency (e.g. the tone emitted from the speaker) will always be 750 Hz.
- Note: There will be no tone in SSB-TUNE mode when frequency translation is active.
- When TUNE is activated in CW mode the frequency offset from the display frequency and the sidetone frequency will be that configured as the sidetone frequency in the menu.
- Pressing-and-holding the TUNE button will toggle the Transmit Disable function. If this is activated the TUNE indicator above button F5 will be displayed in grey and pressing it will have no effect. The “Transmit Disable” function may also be enabled/disabled in the configuration menu.

VFO A (or VFO B):

When not in Menu mode, “soft” button **F4**, beneath the display. This button toggles which VFO, A or B, is currently the “Active” VFO. 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.

If **SPLIT** mode is active the currently active 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.

If one **PRESSES AND HOLDS** this button (**F4**) the currently active VFO's mode, filter setting and frequency are copied to the inactive VFO with an on-screen indication that this has taken place.

SPLIT:

When not in Menu mode, “soft” button **F3** toggles “SPLIT” mode on and off.

When SPLIT mode is off the radio behaves normally, using the *currently selected* VFO for both receive and transmit.

When SPLIT mode is on, the radio uses the currently-selected VFO's mode for **both** 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:

- Activate the SPLIT function. “SPLIT” has now changed color and the display 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.
- Dial in 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.

- Dial in 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:

- When installing and then using this firmware for the **first time** there may be a problem with saving the VFO A/B frequencies. After using the POWER button to save the settings once or twice it appears as though the memory locations get properly initialized and that they work as they should thereafter.
- 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.

“Soft” buttons in normal 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.

“Soft” buttons 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** (button **F1**) – This exits the menu system, returning to the main display. Pressing-andholding this button will save all settings to EEPROM.
- **DEFLT** (button **F2**) – This button resets the currently-selected item to its default setting.
- **PREV** (button **F3**) – This button goes backwards one screen or six menu items. Pressing-andholding this button will jump to the beginning of the menu, or to the end of the menu if already at the beginning.
- **NEXT** (button **F4**) – This button goes forwards one screen or six menu items. Pressing-andholding this button will jump to the end of the menu, or to the beginning of the menu if already at the end.
- The **TUNE** mode remains present while in the **MENU** system at button **F5**

Note: If an item has been changed in the menu system that may need to be saved to EEPROM using the **POWER** button, the **MENU** indicator will be orange and be followed by an asterisk (e.g. “MENU*”)

Configurable options on the main 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” (a.k.a. “Volume Control”). This is used to adjust the audio level feeding the speaker/headphone jack using encoder **ENC1**. Button **M1** may be used to select whether this encoder adjusts **AFG** or **STG** (*see below*) with the “un-selected” item being “grayed” out. **AFG** (e.g. the “Volume control”) 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 using encoder **ENC1**. Button **M1** may be used to select whether this encoder adjusts **STG** or **AFG** with the “unselected” item being “grayed” out. Sidetone Gain is also adjustable from the main menu. *When not in CW mode this is replaced with “CMP”.*

- **CMP** - “TX Compression Level” *while not 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. Button **M2** 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 button **M2** 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. Button **M2** may be used to select whether this encoder adjusts **NB** or **RFG** with the “un-selected” item being “grayed” out. Pressing-and-holding button **M2** 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 also be adjustable from the main menu. *When not in CW mode this is replaced with “MIC” or “LIN”.*

- **MIC** or **LIN** - “Microphone Gain” or “Line Input Gain” *when not 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:

Button **G2** 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 button **G2** 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 is 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 that you are trying to remove!

The "strength" may also be set using button **M2** and **ENC2** without having to enter the menu system. To do this:

- Enable DSP "NR" mode by pressing button **G2**.
- Press button **M2** so that the highlighting changes from **RFG** to **DSP** on the screen.
 - If "**NB**" is displayed instead, press-and-hold button **M2** to change it.
- With **DSP** highlighted, **ENC2** will now allow adjustment of 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 (accidentally!) 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 NOT 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 HS1 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 Setting.

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 HS1 Transceiver – Brief Overview

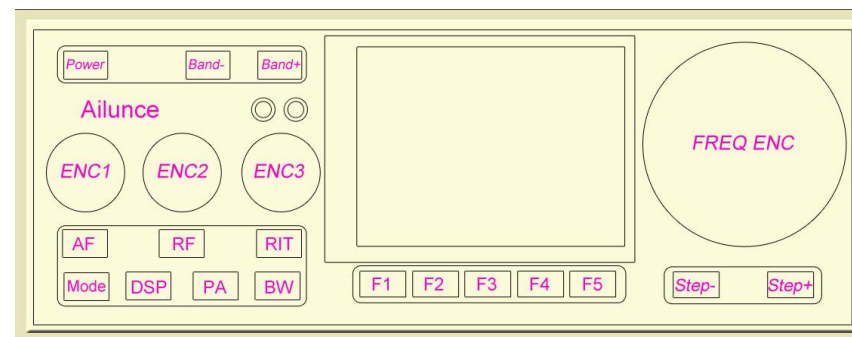


Figure 5: Front panel controls of the HS1 transceiver

Note:

If you are unfamiliar with the HS1, please refer to the section of the manual: “**Before you get on the air - Initial set-up of the HS1 transceiver**”

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 using the **ENC1** control.
- Tune the frequency using **FREQ ENC** knob. Select the step size using the **STEP-** and **STEP+** buttons.
 - Pressing-and-holding the **STEP-** or **STEP+** button will *temporarily* decrease/increase the step size while tuning, the step size display changing colour while this is in effect.
 - Pressing-and-holding both the **STEP-** and **STEP+** buttons at the same time will enable/disable the “Frequency Lock” mode. The

main frequency display will turn grey when “Frequency Lock” is enabled. *The **RIT** is still enabled when the frequency is locked.*

- Change the band using the **BND-** and **BND+** buttons.
- Change the mode (*USB, LSB, CW, etc.*) using button **G1**. *Note: Pressing-and-holding this button will force the selection of “disabled” modes.*

- Button **G4** selects the receiver bandwidth.

Note: Pressing-and-holding this button will force the selection of “disabled” bandwidths.

- Pressing button **G2** will select the mode of DSP noise reduction.

- **Pressing and holding** button **G2** will turn DSP off, 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 **ENC3** 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 the receive frequency and mode, setting the desired output power using button **G3**. ***Note that it is recommended that for voice modes that “full” power not be used unless you have carefully configured for clean, linear output power.***

Initial SSB transmit audio set-up:

- Preferably, connect the HS1 transceiver to a 50 ohm dummy load capable of handling at least 10 watts. Alternatively, you may tune to a **clear** frequency while connected to an antenna with a *known-good* 50 ohm match.
- Use button **G1** to select LSB or USB mode as desired.
- Press button **F2** to select the **AUD** io meter.
- For testing, press button **G3** to select the 0.5 watt setting: The power setting does not matter for this configuration.
- Connect the microphone to connector **J3**: This is the one just above the speaker connector on the right side of the UI board, below and to the right of the **FREQ ENC** control. The HS1 is typically used with an electret-type microphone element and power for the microphone element is supplied by the radio.
- Press button **M3** 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-and-hold button **M3** to change it to **MIC**. Press button **M3** 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 **AUD**io meter bounce upwards. While speaking, adjust the **ENC3**, which adjust the **MIC** parameter, so that the **AUD**io 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 button **F2** to select the **ALC** meter.
- Press button **M1** 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** to a higher value will increase the aggressiveness of the speech processor: A value of **2** is a nice, modest value and a value of 12, while very “punchy” and 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 key your microphone:

- In a quiet room with an antenna or dummy load connected to the HS1, 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

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 must 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 "HS1 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 HS1 with computer “Sound Card” (e.g. digital) modes via the Line-Input and Line-Output connections:

The HS1 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 button **G1**, 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 using **ENC3**: Press button **M3** as necessary to highlight **RIT** 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 using **ENC1**: Press button **M1** as necessary to highlight **CMP** to allow adjustment. 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 (J2) of the HS1 to the audio output of the device you are using to generate the audio and connect the Line-Output jack (J1) of the HS1 to the audio input of that same device.

To key the transceiver, you will need also to connect a cable the Microphone jack (*J3 on the UI board*) **or** the Key jack (*J2 on the RF board*) and the PTT/Key line on either of those jacks (*the “ring”*) would be grounded to key the transceiver: Typical rig-computer interfaces will easily accommodate this connection.

- Preferably, connect the HS1 transceiver to a 50 ohm dummy load capable of handling at least 10 watts. Alternatively, you may tune to a **clear** frequency while connected to an antenna with a *known-good* 50 ohm match.
- Using button **M3**, select **LIN** mode. You may need to press-and-hold this button to change from **MIC** to **LIN**. Press button **M3** as necessary to highlight **LIN**.
- Using button **F2** select the **AUD**io meter.
- Using button **G3** set the HS1 to 0.5 watts for this setup.

- 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 via **ENC3** for a reading on the **AUD**io 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”. ***The Line Output level on the HS1 on this version of firmware is fixed.***
- It should be noted the the **LINE OUT** 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 **F1** to store them in memory

TUNE mode:

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: The HS1 transmits and the indicator turns red.
- Press the **TUNE** button again: The HS1 stops transmitting and the indicator turns white.

Comments about the TUNE mode:

- When set to CW mode, when **TUNE** is activated the HS1 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 HS1 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 gray and all transmit capabilities of the HS1 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 HS1 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

- 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 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 button **G1** to select the CW mode.
- Press button **G4** to select the desired receive audio bandwidth.
- Press button **G3** to set the power to 0.5 watts: The power has little effect on this adjustment.
- Press button **M3** to highlight the **WPM** parameter: Use **ENC3** to set the desired sending speed in words-per-minute. *This parameter has no effect if set to straight-key mode.*
- Press button **M1** to highlight the **STG** parameter: **ENC1** is used to adjust this parameter.
- Press the paddle/key to cause the HS1 to transmit: Use **ENC1** 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 button **G3** when 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**”.

It is recommended that one *NOT* operate CW when the menu is being displayed!

If the menu is being displayed, CW element timing will be disrupted!

The configuration menu system:

The configuration menu may be entered by pressing the **MENU** button (**F1**).

When in the menu system, it may be navigated using the following encoders and buttons:

- **ENC2** – Selects the individual menu item.
- **ENC3** – Adjusts the selected menu item
- **Button F1** – Exits the menu system, returning to the main transceiver display. Pressing-and-holding will save settings to EEPROM.
- **Button F2** – Resets the currently-selected item to its default setting.
- **Button F3** – Goes backwards in the menu system by 6 items (one screen). 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.
- **Button F4** – Goes forwards in the menu system by 6 items (one screen). 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.
- **Button F5** – Enters/Exits **TUNE** mode. Pressing-and-holding this button will also toggle “Transmit Disable”. The “**TUNE**” indicator will turn gray indicating that the transmitter is disabled.

Important Notes:

- When in **MENU** mode **ENC1** is *always* configured as **AFG** (e.g. the volume control.)
- 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 **POWER** button.
- If you have made any changes while in the **MENU** system, when you exit the **MENU** system the label above button **F1** will be orange and display “**MENU ***” to warn you that you should power down using the **POWER** button to save any changes that you might have made.

There are two 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 **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/modif

Main Menu configuration items:

Important Note:

If, when the a menu item is changed, it will be necessary to turn off the transceiver using the **POWER** button to save the changes to the EEPROM.

Alternatively, button **F1** may be pressed-and-held to cause a save of all settings to occur.

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 button **G4** 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 button **G4** 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 button **G4** 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 button **G4** 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 3.6kHz 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 eliminated from selection when the **G1** button is pressed. *Note that it will still be available if one presses-and-holds button G1.*

- **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 >= 10MHz. 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 button **G1** 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 button **G1**: 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 **G1** button. *Note: Even when off, it may still be accessed if button **G1** is pressed-and-held in CW mode. If the AM mode is also set as being disabled, you may have to press-and-hold **G1** 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. “whistleup”) 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 **G4** *while transmitting.*

- **FM RX Bandwidth** – This selects the detection bandwidth when in FM mode: Selection of “audio bandwidth” via button **G4** 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**, **MEDium**, **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.* **WARNING:**

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:

- **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 **AUDio** 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 **AUDio** 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 *this* amount and it *exactly* 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 will 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.

TCXO Related items:

- **TCXO Off/On/Stop** - When set to **OFF** the TCXO is read every

second or so and the temperature is displayed, but the frequency is not corrected based on the temperature. When set to **ON**, temperature-related frequency corrections are applied to minimize frequency drift. When set to **STOP** the temperature sensor is *not* polled and “**STOPPED**” is displayed in lieu of the temperature. The “**STOP**” setting may be used by those who experience the one-second “**TICK**” sound on higher bands (e.g. 15 meters and up) who have not performed the modification to prevent this. **Note:** If you experience this “tick” sound be certain that you have enabled the “**RX/TX Freq Xlate**” mode, preferably setting it to “**RX LO LOW**” before resorting to disabling the TCXO function and losing temperature/frequency control.

- **TCXO Temp. (C/F)** - This selects either Centigrade or Fahrenheit display of the TCXO temperature.

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 slower 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 2x magnify** - When set to **ON** this changes the span of the spectrum scope and waterfall display from its normal +/- 24 kHz to +/- 12kHz. 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, atypically 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 leave it there 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 viceversa. 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 final item of the main menu item is “Configuration Menu”. When set to **ON** the “Configuration” menu is enabled and its menu items are accessible.

General radio setup related items:

- **Step Size Marker** - When set to ON a line below the appropriate digit of the main frequency display indicates the selected step size.

- **Step Button Swap** - When ON, the STEPM (**Step-**) and STEPP (**Step+**) buttons are swapped. The intent of this is so that the position of the **Step Size Marker** moves to the left/right in conjunction with the left/right step size button when this setting is on.

- **Band+/- Button Swap** – When on, the BANDM (**Band-**) and BANDP (**Band+**) buttons are swapped. This is provided for those who wish to these buttons to be swapped – perhaps, because they also

have the **STEP** buttons swapped as well.

- **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 batterypowered. When the LCD backlight is turned off, the spectrum scope and waterfall 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 button **G3** – *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 button **G3** 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 NOT 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. Calibrate** - This adjusts the frequency calibration of the transceiver in Hz, referenced to 14.000 MHz. Please refer to the section “**Calibrating the HS1’s Operating Frequency**” at the end of this document.
 - Use the **STEP-** and **STEP+** buttons - changing the step size as necessary - when making this adjustment. Note that if the **STEP** buttons are used, only up to 1 kHz step sizes will be selectable, but if a larger step size was set *prior* to entering this menu function, that step size will be used.

- **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 Si570, 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

- **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.

PA bias adjustment related items:

- **CW PA Bias (If >0)** - If this setting is greater than zero, it is sets the applied PA bias during transmit when in CW mode. If this value is zero the setting of “**PA Bias**” (see below) is used.
- **PA Bias** - This is the setting applied to the final transistors during transmit. If the value of “**CW PA Bias**” is set to a value of zero, this value is used during CW transmit.
 - The signal gain of FET power transistors will vary with their bias. If you set the PA bias in CW lower than that in SSB mode, you can expect that the RF power output will be **LOWER**. The converse will be true if the PA bias is set higher (e.g. higher power

Note:

- It is not possible to enter CW transmit mode without RF drive being present. If it is desired that the PA bias current be measured for CW

transmit, it is necessary that the bias first be set in SSB mode, **NOT** in **TUNE** mode, by keying the PTT with no audio and then noting the numerical value at the desired PA current. Once this value has been obtained, it may be applied to the **CW PA Bias** setting.

Power/VSWR Meter related items:

- **Disp. Pwr (mW)** – When set to **ON** this enables the display of forward and reflected RF power, in that order, in milliwatts in the upper-left corner, just below the “RIT” and “WPM” displays and is updated **only** when the transmitter is keyed. In order to remove these numbers from the screen, set this to **OFF** and key the transmitter.

◦ *Note that this setting is NOT saved in EEPROM and is always OFF upon power-up.*

- **Pwr. Det. Null** – This nulls the forward and reverse power sensors **when no RF power is present**. This setting is enabled **ONLY** when “**Disp. Pwr (mW)**”, above, is enabled. This setting is to be adjusted **ONLY** when the transmitter is keyed in SSB mode **WITH NO MICROPHONE CONNECTED** and is just to the point that the forward and/or reverse power indicators **just** flicker between 0 and 1 or 2.

◦ **Note:** If the resistor-change modification to the SWR circuitry has not been done you will not be able to “null” this out and you will get a warning to this effect when you power-up the transceiver.

- **80m Coupling Adj.** - This adjust the calibration factor for the forward and reverse power sensors when operating on 80 meters. If a known-accurate wattmeter is available, use this adjustment with “**Disp. Pwr (mW)**” enabled to accurately set the wattmeter for this frequency range. A setting of 100 represents unity (1.00).

- **40m Coupling Adj.** - This adjust the calibration factor for the forward and reverse power sensors when operating on 40 and 60 meters. If a known-accurate wattmeter is available, use this adjustment with “Disp. Pwr (mW)” enabled to accurately set the wattmeter for this frequency range. A setting of 100 represents unity (1.00).
- **20m Coupling Adj.** - This adjust the calibration factor for the forward and reverse power sensors when operating on 20 and 30 meters. If a known-accurate wattmeter is available, use this adjustment with “Disp. Pwr (mW)” enabled to accurately set the wattmeter for this frequency range. A setting of 100 represents unity (1.00).
- **15m Coupling Adj.** - This adjust the calibration factor for the forward and reverse power sensors when operating on 17, 15, 12 and 10 meters. If a known-accurate wattmeter is available, use this adjustment with “Disp. Pwr (mW)” enabled to accurately set the wattmeter for this frequency range. A setting of 100 represents unity (1.00).
- **FWD/REV ADC Swap** - This swaps the A/D inputs of the forward and reverse power detectors. This may be useful if the builder reconfigures the FWR/REV coupler such that its detection sense is reversed, such as by rewiring it back-to-front or changing the “sense” of the windings. This may be done if the builder finds that the reverse isolation of the unit's “Tandem” coupler is better when so-configured.

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:

$$\text{Displayed Frequency} = (\text{Tuned Frequency} * \text{XVTR Mult}) + \text{XVTR Offset}$$

5 Watt Power calibration items:

These are accessible **ONLY** if the transciever is set to the relevant band **AND** the 5 watt setting:

- **80m 5W PWR Adjust** - This adjusts the drive level so that 5 watts may be obtained on this band.
- **60m 5W PWR Adjust** - This adjusts the drive level so that 5 watts may be obtained on this band.
- **40m 5W PWR Adjust** - This adjusts the drive level so that 5 watts may be obtained on this band.
- **30m 5W PWR Adjust** - This adjusts the drive level so that 5 watts may be obtained on this band.
- **20m 5W PWR Adjust** - This adjusts the drive level so that 5 watts may be obtained on this band.
- **17m 5W PWR Adjust** - This adjusts the drive level so that 5 watts may be obtained on this band.
- **15m 5W PWR Adjust** - This adjusts the drive level so that 5 watts

may be obtained on this band.

- **12m 5W PWR Adjust** - This adjusts the drive level so that 5 watts may be obtained on this band.
- **10m 5W PWR Adjust** - This adjusts the drive level so that 5 watts may be obtained on this band

Important notes on the 5 watt power adjustment settings:

- If you see the settings for the 5 watt power adjust defaulting to ZERO, first POWER OFF the radio using the POWER button to initialize the memory-save locations.
- If you do not have the 5 watt mode *AND* the band to be adjusted selected, the relevant parameter will be “oranged out” and you will *not* be able to adjust it – this being done to prevent accidental adjustment of the wrong parameter.
- The 2 watt, 1 watt and 0.5 watt levels are based on proportional scaling of the 5 watt settings.
- While you may get 5 watts in TUNE or CW mode, your *measured* output power may be lower in SSB mode owing to the peak-average nature of SSB. *Unless* you have a peak-reading SSB power meter, *do not* trust it to properly read the power output when in SSB! Also, remember that the adjustment of the **Mic Gain** (or Line Gain) settings will affect your output power when in SSB.
- **NOTE** that unless your final/driver amplifier is appropriately modified, you may not be able to get full 5 watts on some of the higher bands (e.g. 15 meters and above.) *Please follow the discussions on the Yahoo Group and check the “HS1 board modifications” document for updates on this topic.*

“FULL” Power calibration items:

These are accessible ONLY if the transceiver is set to the relevant band **AND** the FULL power setting:

- *80m FULL PWR Adjust* - This *adjusts the drive level so that “full” linear power may be obtained on this band.*
- *60m FULL PWR Adjust* - *This adjusts the drive level so that “full” linear power may be obtained on this band.*
- *40m FULL PWR Adjust* - *This adjusts the drive level so that “full” linear power may be obtained on this band.*
- *30m FULL PWR Adjust* - *This adjusts the drive level so that “full” linear power may be obtained on this band.*
- *20m FULL PWR Adjust* - *This adjusts the drive level so that “full” linear power may be obtained on this band.*
- *17m FULL PWR Adjust* - *This adjusts the drive level so that “full” linear power may be obtained on this band.*
- *15m FULL PWR Adjust* - *This adjusts the drive level so that “full” linear power may be obtained on this band.*
- *12m FULL PWR Adjust* - *This adjusts the drive level so that “full” linear power may be obtained on this band.*
- *10m FULL PWR Adjust* - *This adjusts the drive level so that “full” linear power may be obtained on this band.*

Important notes on the “FULL” power adjustment settings – PLEASE READ CAREFULLY:

- If you see the settings for the “FULL” power adjust defaulting to ZERO, first POWER OFF the radio using the POWER button to initialize the memory-save locations.
- If you do not have the FULL power mode AND the band to be adjusted selected, the relevant parameter will be “oranged out” and

you will not be able to adjust it – this being done to prevent accidental adjustment of the wrong parameter.

- The “FULL” power setting has NO effect on any other power setting.
- While you may note a certain power output in TUNE or CW mode, your measured output power may be lower in SSB mode owing to the peak-average nature of SSB. Unless you have a peak-reading SSB power meter, do not trust it to properly read the power output when in SSB! Also, remember that the adjustment of the Mic Gain (or Line Gain) settings will affect your output power when in SSB.
- Note that “officially” the HS1 transceiver is just a 5 watt radio, but work is being done to derive modifications to safely increase the output power.
- It is recommended that you do NOT increase the output power above 10 watts unless you have verified that you have provided adequate heat sinking of the final power transistors.
- Because the gain of the circuitry decreases with increasing frequency, you should expect that the maximum power will decrease on the higher bands! This is not a malfunction, but the reality of semiconductor physics!
- If the output power is set too high, nonlinearity may result, causing key clicks on CW and “splatter” on SSB and/or AM, so please take care when adjusting the “FULL” power parameters!
- *Please follow the discussions on the Yahoo Group and check the “HS1 board modifications” document for updates on the topic of improving the power amplifier of this radio!*

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 “decorrelate” 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 larger 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”. *A higher number will increase processor loading and slow the user-interface response.*

- **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 *impression* 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:

- **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, ***PLEASE be considerate*** 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):** 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 by 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 is 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 *always disabled* 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 codec AGC is still active and the receiver may still desense 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.

There are some known problems. The DSP (especially the noise reduction) tends to crash occasionally:

- Occasionally, the receiver audio will suddenly cut out when in Noise reduction and/or Notch mode. As of version 0.0.214 there is an algorithm that automatically detects most of the situations when this occurs and will reset the DSP. In the event that a crash is not automatically detected, you may reset the DSP by turning it (the DSP) off and on again by either pressing, or pressing-and-holding button **G2**.

Important Note:

If, when the a menu item is changed, it will be necessary to turn off the transceiver using the POWER button to save the changes to the EEPROM.

Alternatively, button F1 may be pressed-and-held to cause a save of all settings to occur.

Approximate specifications of the HS1 transceiver:

The following specifications are for a transceiver that has been modified according to the “hs1_board_modifications_xxxx” file that may be found in the FILES section of the YAHOO HS1 Yahoo group.

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

- **Receiver 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:** 80, 60, 40, 30, 20, 17, 15, 12 and 10 meter amateur bands, transmit. Receive: 3.5-30 MHz nominal including general coverage, 1.8-32 MHz at reduced specifications.

- Note that the ability of the Si570 to tune the radio below 2.5 MHz is not guaranteed in its specifications, but most units have enough range to tune just below 1.8 MHz.

- **Spectral Display Modes:**

- **Spectrum Scope:** This is a spectrum analyzer with the vertical divisions representing userdefinable 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 userselected dB/division. A graticule along the bottom of the display indicates the approximate frequency of the signal being displayed over a width of +/- 24 kHz. (48 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” mode for both the Spectrum Scope and Waterfall Display mode that provides 2x magnification, reducing the visible spectral width to just +/- 12 kHz (24kHz total.)

- Both the Spectrum Scope and Waterfall Display are very 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.
- **Transmitter power output:** 5 Watts, typical, linear. *Modifications may be made to increase this: Follow the discussions in the Yahoo group.*

- **Frequency stability:** +/- 30 Hz at 14 MHz over the range of 10 to 35 C, ambient with the transceiver in the case or better with the TCXO active. *(It can be much better than this.)*

- **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 softwaredefined 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 (obvious reasons!) or when using a “wide” receive bandwidth.*

- **S-Meter calibration:** “Industry Standard” *(IARU Region 1, Technical recommendation R.1)* Smeter 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 input/output connections:** “Line In” and “Line Out” audio ports, and a “PTT” (Push-to-Talk) are provided via 3.5mm 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 HS1 such as SSTV, PSK31, WSPR and other analog/digital modes.

- **Line out signal levels:** Nominal 1 volt peak-peak, maximum when AGC is operating.

- **Line in signal levels:** Nominal 0.1-1.0 volt peak-peak, adjustable using the “Line Input Gain” settings.

- **Transmit ALC type:** Look-ahead gain compressor with both pre-set and available “custom” settings.

- **Operating voltage range:**

- 16.0 volts maximum
- To 11.0 volts minimum for full transmit power
- To 9.5 volts for reduced transmit power
- As low as 6.5 volts, receive-only: Only very low transmit power may be possible – distortion on peak audio (SSB, AM) may occur.

- **Current consumption:**

- **Receive:**
- Unmodified, approx. 410mA on 40 meters and below at 13.0 volts, approx. 440 mA on 10 meters, minimum volume, maximum display brightness.
- The selection of minimum 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:** <5 mA if the PA driver bias modification is performed. *(If this modification is not performed it is recommended that power be disconnected from the transceiver as the PA driver transistors will be biased on even when the power is turned off, causing 30-60 mA of standing current.)*

The ALC (Automatic Level Control)

This module requires a bit of explanation, so please read the following section very carefully!

Prior to the addition of the ALC the **POWER** adjustment on the HS1 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 the **F2** button has been repurposed to change the (former) **SWR** meter to one of three modes:

- The **SWR** meter. This dynamically measures the forward and reflected RF power, calculates the VSWR and displays it.
- The **AUDio** 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.

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

- 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 **AUDio** 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 **AUDio** 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 never increase it and it will settle to unity under nosignal conditions. *Note that the ALC meter's response is 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 of 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:

- 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.
- Frequent, very high indications on the **ALC** meter (*e.g.* $>12\text{dB}$) can cause annoying “pumping” of background noise on transmit audio, which is to say that during periods of silence in the voice, sounds in the background may rise up and become an annoyance to those listening to the transmission on the air. A “fast” ALC release time (*e.g.* *low number*) can make this effect worse.
- The use of a speech compressor/processor can significantly increase the heat dissipation of the final transistors: Please be aware of this while transmitting, making sure that your finals are adequately heat-sinked!
- If the HS1 is being operated from lower than normal supply voltage the RF amplifier may be unable to output a normal amount of power. In severe cases, operating under these conditions may result in distortion and/or “splattering” which can cause interference on the
- If operating at “**FULL**” power, splattering may result unless you had carefully adjusted the “**FULL Power**” configuration settings such that the obtained power level was within the linear range.
 - **If the “FULL” power setting was simply adjusted for maximum power output you may expect that transmissions at this power setting may sound somewhat distorted and could cause a degree of “splattering”.**

SSB operation and proper adjustment of the “5W PWR Adjust” and “FULL PWR Adjust” parameters”:

If you get reports of “splattering” when you operate on SSB, first check the **AUD**io meter to make sure that it is not “buried” in the red, then check the **ALC** meter to verify that the **CMP** (“TX Compress Level”) is not set such that this meter indicates excessive deflection in the red zone (*e.g.* continuous excursions above 12-16dB). If you have appropriately adjusted the microphone/line and ALC settings, but are still getting reports of splattering, do the following:

- If you are on “**FULL**” power, set the power to **5 Watts**.
- If you are running **5 Watts**, set the power to a lower setting.
- If you are still getting reports of splattering, verify that the **PA Bias** is properly set.

If reducing power “cleans up” the splattering problem, your final power amplifier may not be able to output the expected amount of power on the current amateur band and this could be for a number of reasons:

- **The power supply voltage is low.** If you are operating the radio on a voltage lower than 12.5 volts, it may not be able to output power level that you request so a lower power level should be selected if you operate at that voltage.
- **There may be a problem with the low-pass filter on that band.** You should compare the output power on that band with that of other bands and if it is markedly lower, re-check – and readjust, if necessary – the values of the toroidal inductors in that band's low-pass filters.
- If you get reports of splatter on “**FULL**” power but not on other power settings, you should reduce the “FULL PWR Adjust” for that band in the “Adjustment Menu” for that band, or remember to *not* operate SSB at the “**FULL**” power setting. Remember: It is possible to get quite a bit of RF output from the final transistors, but if it is **Linear** and “clean” power that you want – necessary for SSB operation – you

will need to operate at a lower power than this “maximum” output!

- If the **PA Bias** was never set properly your RF amplifier, nonlinear operation may result. In order to set the **PA Bias** do the following steps:

- Adjust the radio's power supply for 12.5-14.0 volts.
- Insert an ammeter so that you can measure its current consumption. This meter should be capable of reading up to 3 amps with a resolution of better than 0.1 amps.
- Connect the radio to a dummy load.
- Turn on the radio and select USB or LSB.
- Go to the adjustment menu and set the **PA Bias** to **0** (zero).
- Key the radio *with no audio* and note the current.
- With the radio keyed, adjust the **PA Bias** so that the current *increases* by the desired bias current: Up to 0.5 amps is suggested, although as low as 0.1 amps is adequate for linearity if you obtain the desired amount of RF output power on all bands.
- Unkey the radio.
- Use the **POWER** button (or press-and-hold the MENU button) to save the new **PA Bias** setting.

REMEMBER:

If your RF power meter does not have a good "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:

As of firmware version 0.0.217 there is the ability to enable “Frequency Translation” which mathematically shifts the center frequency by + or – 6 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 6 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 always 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 ± 2 kHz 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.

How the AM demodulation in version 0.0.208 and later is different from that in earlier versions:

In versions prior, the *pre-demodulation* bandwidth was fixed at 10 kHz, which meant that all signals within ± 10 kHz would hit the demodulator. Because the demodulator is, by its nature, a non-linear “device” it would mix and cause distortion should any other signal within that ± 10 kHz

passband also be intercepted. The selectable audio filter was applied *after* the demodulation, but if there was an extraneous signal within the ± 10 kHz passband, the damage was already done!

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 lowpass cut-off frequency selected according to the desired bandwidth. Post-detection, there is

The “Zero-Hertz” hole problem if using firmware earlier than 0.0.217 or you are using firmware 0.0.217 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 terribly distorted – much like an SSB signal tuned on an AM receiver!**

The cure for this 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!

Known issues with AM demodulation if using firmware *earlier* than 0.0.217 or you are using firmware 0.0.217 with “Frequency Translation” disabled:

- **Remember:** AM signals *must* be off-tuned to avoid placing the carrier in the dead center! An offset of a few hundred Hz is typically

adequate.

- There is a known issue in which a weak heterodyne (“*tweet*”) can be heard at frequencies close to the center frequency. This is caused by the inexact 90 degree phase shift and slight amplitude imbalance in the receive system: For the time-being, off-tune the carrier until it disappears *or* you may try turning on the DSP notch filter, and you should adjust the parameter “**AM RX IQ Bal.**” to minimize it.

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 lowlevel “*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.

A recommended modification for HS1 Board version 0.4 (and possibly earlier) if you are interested in AM reception:

As noted in the modification file it is recommended that capacitors **C71** and **C73** (on the outputs of U16 of the RF board) be removed and replaced with jumpers or zero-ohm resistors: Their DC-blocking function is provided by capacitors C26 and C31 on the UI board and the removal of C71 and C73 will extend the low-frequency response of the receiver and reduce the width of this “hole” significantly. It also has the side-effect of potentially improving the low-frequency opposite sideband rejection as it is one-fewer

component in the audio path to have its value change with temperature and cause a phase/amplitude shift.

AM Transmission:

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:

- **As of this firmware version the FM mode is new and is disabled by default.**
- **Expect there to be bugs and “rough edges”!**
- **Your mileage may vary, so there!**
- **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 HS1 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 HS1 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 HS1 is compatible with PM, which is to say that for modulation, a 6dB/octave preemphasis 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 ENC2. 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 some strange ones) 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:

Subaudible 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 ***Press-and-Hold*** button **G4**, the “Bandwidth Change” button.
- 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, 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.

**Before you get on the air -
Initial set-up of the HS1 transceiver:**

- **Be certain that you have reviewed the “HS1 board modification” file.** This document details various modifications that will improve performance of the HS1 and prevent problems.
- **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.** This does not default to “ON”, but there is a warning that is displayed on power-up if it is not activated. Be certain that the adjustment menu (*yellow*) parameter “RX/TX Freq Xlate” is set to “RX LO LOW”.
- **PA Bias setting:** Before RF output may be obtained, the PA bias must be properly set as follows:
 - **DO NOT PERFORM THIS STEP UNLESS THE RF OUTPUT TRANSISTORS ARE PROPERLY HEAT-SINKED!**
 - Connect an ammeter in series with the HS1 power supply. This should be capable of 3 amps with better than 0.1 amp resolution.
 - Connect a known-good dummy load to the HS1.
 - Set the HS1 to 10 meters and the mode to LSB or USB.
 - Go to the Calibration menu item “PA Bias”.
 - With **NO AUDIO** present (e.g. minimum microphone gain and quiet room) key the transmitter: **QUICKLY** set the bias to ZERO – this may be done by pressing the **DEFLT** button (**F2**). *Note the current reading on the ammeter.*
 - With the transmitter still keyed, increase the **PA Bias** setting so that there is a 0.5 amp increase in the current reading on the ammeter, which should correspond with 0.25 amps per RF output transistor. Once you have obtained this, unkey the transmitter.
 - Press-and-hold **F1** to save the setting to EEPROM.

◦ **Important notes:**

- Depending on the RF drive level and transistor gain, the bias setting may need to be adjusted slightly: The more bias, the higher the gain and vice-versa. You should not need more than 1 amp, hopefully!
- The “**CW PA Bias**” adjustment is specifically for CW operations and is described earlier in this manual.

- **Band-Gain adjustments:** This sets up the transmitter gain on a per-band basis according to the following procedure:

◦ **DO NOT PERFORM THIS PROCEDURE UNLESS THE RF OUTPUT TRANSISTORS ARE PROPERLY HEAT-SINKED!**

- If you have already verified that the HS1's built-in wattmeter is properly calibrated, you may use it for the following steps. If you do so, set the parameter “**Disp. Pwr (mW)**” to **ON** to display the transmitted power in **MILLIWATTS** (e.g. watts * 1000).
- Connect a power supply capable of at least 3 amps while supplying between 12 and 14 volts.
- Connected a known-good wattmeter and dummy load.
- Set the transmitter to CW mode.
- Set the band to 80 meters.
- Set the power to 5 watts.
- Go to calibration menu item “**80m 5W PWR Adjust**”.
- Press the **TUNE** button to enter transmit, a tone should sound in the speaker: The number to the right of the “**80m 5W PWR Adjust**” should turn white.

- If the number does not turn white, press **TUNE** again to return to receive mode and verify the band and 5 watt power

◦ Adjust the parameter for “**80m 5W PWR Adjust**” upwards and note if you get at least 7 watts. If you do, adjust this setting for 5.0 watts output on your external RF wattmeter.

▪ If you are NOT able to obtain at least 7 watts on the band you will not be able to obtain a “clean” 5 watts on SSB, either! It is recommended that the final power setting be no higher than approximately 75% of the maximum, saturated power obtained! *Failure to do this downward power adjustment will likely result in clipping/splattering on SSB/AM voice peaks on that band!*

◦ **WRITE DOWN the setting obtained for each band.**

◦ Press the **TUNE** button to return to receive.

◦ Repeat the above steps for each band.

◦ Once the “**5W PWR Adjust**” settings are obtained, set the HS1's power setting to FULL, the band to 80 meters, and go to the “**80m Full PWR Adjust**” setting.

◦ Press **TUNE** to go to transmit.

◦ Increase the setting and note the maximum power setting – then press **TUNE** again to stop transmit, going back to receive.

◦ Calculate the power level that is 75% of the reading you obtained in the step above and *write this number down for that band.*

▪ **WARNING:** Failure to set this at 75% power (e.g. below saturated) will likely result in clipping/splattering on SSB/AM voice peaks on that band!

◦ Press **TUNE** again and adjust the setting to obtain the calculated power.

◦ Press **TUNE** again to stop transmit and go back to receive.

◦ Repeat the above steps for all bands.

◦ Now, review again the values stored for the “**5W PWR Adjust**” settings for each band.

▪ None of the band-gain values should be higher “75%” values that you calculated for the “**Full PWR Adjust**” settings for the corresponding bands.

▪ If you set any “**5W PWR Adjust**” parameters higher than then corresponding “**Full PWR Adjust**” for the same band, reduce the “**5W PWR Adjust**” to be *no higher*.

◦ Once you have determined the proper values, press-and-hold the **F1** button to save them to EEPROM.

• **Wattmeter/SWR meter adjustment:**

◦ Connect a dummy load to the transmitter

◦ Make certain that the MICROPHONE input is selected and turn the microphone gain setting to minimum.

◦ Set to USB or LSB.

◦ Set the band to 10 meters.

◦ Set the power to 0.5 watts.

◦ Go to the **CALIBRATION** menu and set the item “**Disp. Pwr (mw)**” to **ON**. This will enable the display of the detected power in milliwatts in a portion of the screen just below the “**RIT**” indicator with first number indicating forward power and the second indicating reverse power.

◦ With NO audio present (e.g. a quiet room or keying the transceiver with no microphone present) observe numbers in the upper-left corner of the screen, just below the “**RIT**” indicator, and adjust the calibration menu setting “**Pwr. Det. Null**” to zero these numbers.

◦ Once the **NULL** value has been determined, un-key the

transmitter and then press-and-hold the **F1** button to save the setting to memory.

- It is possible that these two numbers will not exactly coincide at zero, but they should be within less than 5 units of each other. These numbers will also vary slightly in a random fashion: It is better that these numbers occasionally go slightly above zero than always stay below zero.

- If the forward and reverse power are more different than 5 or so points, check the following:

- Make sure that there is no audio present – not even noise! There must be NO transmit power in this step. If necessary, set “**10m 5 W PWR Adjust**” setting for the current band to the lowest possible number, noting the original setting: You will have to momentarily set the power to 5 watts to allow this adjustment – but remember to set it back to 0.5 watts. Once you are done, restore the original “**10m 5 W PWR Adjust**” value and then press-and-hold **F1** when done to save the setting.

- If the forward/reverse readings are still different, make certain that you have installed the proper-value components during the modification.

- If resistors of lower precision than 1% were used (e.g. 5%) it is possible that normal tolerance variations are responsible for these differences, in which case it may be necessary to use a digital ohmmeter to match several pairs of resistors to within a fraction of a percent.

- Check to see if your solder is based on “Water soluble” flux: If it is, be absolutely certain that this flux is removed as it can become conductive upon absorbing moisture from the air. This type of flux is more common with “Lead Free” solders.

- If you still get radically different forward/reverse readings, check the voltages on pins 9 and 10 of the 30 pin connector: They should be equal, within a few millivolts. If not, check wiring.

- After functioning of the VSWR meter is verified and the **OTHER** VSWR/Power meter (e.g.T2/T3) related modifications discussed in this document have been performed, the wattmeter may be calibrated as follows.

- ***DO NOT PERFORM THESE STEPS UNLESS YOU HAVE PROPERLY HEATSINKED THE FINAL TRANSISTORS!***

- Connect a known-good 50 ohm dummy load and wattmeter to the output of the HS1.

- Set the transmit power to 5 watts.

- Set the mode to USB, LSB or CW.

- Go to the “**Pwr. Det. Null**” item and set it to **ON**. This will cause the forward and reverse power to be displayed just below the **RIT** indicator, in **milliwatts**.

- Go to the “**80m Coupling Adj.**” item in the configuration menu.

- Press the TUNE to activate the transmitter.

- Quickly adjust the “**80m Coupling Adj.**” parameter so that the displayed forward power in milliwatts matches your wattmeter's power reading. (Remember: 5000 milliwatts = 5 watts.)

- Repeat the above steps for 40 meters using the “**40m Coupling Adj.**” parameter.

- Repeat the above steps for 20 meters using the the “**20m Coupling Adj.**” parameter.

- Repeat the above steps using the “**15m Coupling Adj.**”, **using a frequency on 12 meters.**

- Once you are done setting these values, press-and-hold the F1 button to save the settings to EEPROM.

• **Frequency Calibration:** This step requires several things:

- 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 HS1 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 ENABLED:

1. Set the HS1 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 HS1 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 HS1 transceiver 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 HS1 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 HS1 dial frequency 1 kHz above the carrier frequency to obtain a strong 1 kHz audio note.
- 5) Now tune the HS1 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/Smeter 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 HS1 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 sidetone 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).

Recommended procedure for adjusting TX IQ gain and phase adjustments

Frequency translate mode ENABLED:

1. **Loosely** couple your transmitter to a receiver that is connected to a computer running a program with a waterfall display. Do NOT connect the HS1 transmitter to your receiver, but *connect it to a dummy load* and place an antenna coupling from your computer-connected receiver nearish the HS1's RF output so that it gets adequate signal. A program such as **Spectran** is recommended. *(The **Spectrum Lab** program will also work, but is more complicated to use.)*
2. Switch to LSB mode on the HS1.
3. Set the HS1 to 1 Watt mode.
4. Switch to **USB** mode on the computer-connected receiver. *(Yes, USB.)*
5. Tune both the HS1 and the computer-connected receiver to the same frequency.
6. Press the TUNE button. The audio on the HS1 audio should mute and you should see, on the waterfall display, a signal 750 Hz below the HS1 dial display frequency. *If the dial frequency is 7200 kHz, the main LSB signal will be on 7199.25 kHz, 750 Hz below 7200 kHz.*
7. Now look at the opposite sideband signal (the USB image) at 7200.75 kHz and adjust "**LSB TX IQ Phase**" to minimize it. There will likely not be much difference in this signal, perhaps 2-10 dB. If there is no easily discernible difference using your measuring method set this value to zero.
8. Now, adjust "**LSB TX IQ Bal**" for maximum suppression at the frequency of the *translated image*. This signal will be located approximately 12 kHz away from the dial frequency. You should Enable/Disable TUNE mode and/or adjust the setting of "**LSB TX IQ Bal**" to positively identify this image signal.

Note that this adjustment may not null it completely out, but there should be an obvious minimum setting.

- If the frequency translate mode is set to "**RX LO Low**" find a signal **11.25 kHz below** the dial frequency. *In the example where the dial frequency is 7200 kHz, this signal would be found at 7188.75 kHz.*

9. Press TUNE again to exit transmit mode and switch to USB mode, the computer-connected receiver to LSB, and press TUNE to produce a transmit test signal.
10. Press the TUNE button. The audio on the HS1 audio should mute and you should see, on the waterfall display, a signal 750 Hz above the HS1 dial display frequency. *If the dial frequency is 7200 kHz, the main USB signal will be on 7200.75 kHz, 750 Hz above 7200 kHz.*
11. Now look at the opposite sideband signal (the LSB image) at 7.19925 MHz and adjust "**USB TX IQ Phase**" to minimize it. There will likely not be much difference in this signal, perhaps 2-10 dB. If there is no easily discernible difference using your measuring method set this value to zero.
12. Now, adjust "**USB TX IQ Bal**" for maximum suppression at the frequency of the *translated image*. This signal will be located approximately 12 kHz away from the dial frequency. You should Enable/Disable TUNE mode and/or adjust the setting of "**USB TX IQ Bal**" to positively identify this image signal. Note that this adjustment may not null it completely out, but there should be an obvious minimum setting.

- If the frequency translate mode is set to "**RX LO Low**" find a signal **12.75 kHz below** the dial frequency. *In the example where the dial frequency is 7200 kHz, this signal would be found at*

7187.25 kHz.

◦ If the frequency translate mode is set to “**RX LO High**” find a signal **11.250** kHz *above* the dial frequency. *In the example where the dial frequency is 7200 kHz, this signal would be found at 7211.25 kHz.*

13. Press **TUNE** again to return to receive mode.

14. Repeat the above steps for both AM and FM mode using the parameters “**AM TX IQ Bal.**” and “**FM TX IQ Bal.**”, respectively.

◦ **NOTE:** In both AM and FM mode the TUNE function will not work: Simply key the transmitter and use the constant carrier produced by those modes as your test signal.

Important:

- When you are done write down the phase and gain settings, then power off using the **POWER** button to save the settings. Power up again and return to the menu to verify that they were saved.

Recommended procedure for adjusting TX IQ gain and phase adjustments

*Frequency translate mode **DISABLED**:*

Do not perform the following procedure unless you operate the HS1 transceiver 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. **Loosely** couple your transmitter to a receiver that is connected to a computer running a program with a waterfall display. Do NOT connect the HS1 transmitter to your receiver, but *connect it to a dummy load* and place an antenna coupling from your computer-connected receiver nearish the HS1's RF output so that it gets adequate signal. A program such as **Spectran** is recommended. *(The **Spectrum Lab** program will also work, but is more complicated to use.)*
2. Switch to **LSB** mode on the HS1.
3. Set the HS1 to 1 Watt mode.
4. Switch to **USB** mode on the computer-connected receiver. *(Yes, USB.)*
5. Tune both the HS1 and the computer-connected receiver to the same frequency.
6. Press **TUNE** mode. You will hear a 750 Hz tone from the HS1 and see it on the waterfall display, see a signal 750 Hz below the HS1 dial display frequency. ***Note: You'll have to do some simple math to figure out where these frequency components will land on the waterfall!***
7. If you adjust the **LSB TX IQ Bal.** you should see the signal 750 Hz above the HS1 dial display frequency go up and down. Null this upper frequency as much as possible.

8. **Note:** Unless the IQ Gain balance is nulled as much as possible, nulling of the Phase adjustment will *not* be possible.
9. Once the best nulling is obtained with the **LSB TX IQ Bal.**, adjust the **LSB TX IQ Phase**. It will "click" on each adjustment, so wait for the waterfall screen to clear after each adjustment.
10. Once the best phase null is obtained, go back and forth between the gain and phase for the best null.
11. Press **TUNE** again to exit TUNE mode.
12. Switch the HS1 to **USB** mode and the computer-connected receiver to **LSB** mode.
13. Switch the computer-connected receiver in **LSB** mode, but this time null out the 750 Hz tone BELOW the HS1 dial frequency, having done the math to figure out where the frequency components will land!

When you are done write down the phase and gain settings, then power off using the **POWER** button to save the settings. Power up again and return to the menu to verify that they were saved.

Calibrating the HS1's operating frequency

The HS1 transceiver has provisions for calibrating the display frequency to that of known-accurate frequency references such as a time station or a frequency reference using the "**Freq. Calibrate**" menu item.

To calibrate the HS1 frequency:

This menu item is adjustable from -9999 to 9999, this representing Hz at a frequency of 14.000 MHz, proportionally affecting all operating frequencies. When making this adjustment, use the **STEP-** and **STEP+** buttons to select the step size: Note that only the 1 Hz, 10 Hz, 100 Hz, and 1 kHz sizes are actually useful considering the +/-9999Hz adjustment range!

If you use the TCXO feature of the HS1:

- **IMPORTANT:** First, make sure that you have thermally bonded the Si570 (U8) and the temperature sensor chip, U10 with a piece of copper or aluminum that is at least 1mm thick. It is recommended that metal-filled (e.g. grey or black) epoxy be used for this, although clear epoxy will also work. It should be noted the RTV (e.g. "Silicone") adhesive does not conduct heat very well and that cyanoacrylate (e.g. "Super") glue works quite poorly as it will likely leave a gap.
- **Set RIT to zero** by adjusting **ENC3** so the small frequency display's frequency matches that of the main (RX) frequency display. (*The RIT always defaults to "zero" on power-up.*)
- If you have a known-accurate frequency counter, couple it to with a capacitor or highimpedance probe to pin 2 or 14 of U15 on the RF board: You should see the receive frequency: Adjust the "**Freq. Calibrate**" parameter so that the frequency on the counter and the HS1 display agree.

- If a sensitive receiver or service monitor is available, it may be possible to detect the local oscillator radiating from the HS1 and observe the oscillator frequency directly. If you do this, *temporarily* set the “**RX/TX Freq Xlate**” setting to **OFF** so that the local oscillator frequency is the same as the displayed frequency. If you don't set it to “off” note that the local oscillator will be exactly 6 kHz above or below the displayed frequency in receive, depending on the setting of this parameter.
- If adjusting it using the "zero-beat" method on receive, set the bandwidth to 3.6 kHz or to a “wide” filter and tune in a known-accurate signal such as WWV. When setting the frequency, switch between USB and LSB to verify that there is no difference in the way it sounds. The use of one of the wider filters (≥ 3.6 kHz) and the extended low frequency response makes zerobeat adjustments easier, particularly if headphones are used.
- If you have a known signal source and a way to accurately measure audio frequency, tune in that signal source, offset by a given amount (say, 1 kHz) and then adjust the HS1 for a precise 1 kHz audio tone. A computer running a sound card audio analysis program such as Spectran or Spectrum Lab – or even a PSK31 program – can yield accurate frequency information, but be aware that many computers (*particularly laptops or so-called “netbooks”*) can have inaccurate sample rates that cause erroneous frequency measurements. Before you trust the computer to give an accurate frequency reading, it is recommend that you feed audio from a known-accurate source and check calibration. One source of such a signal is the audio from WWV or WWVH (*tuned using AM – never SSB!*) that transmit tones of 500 or 600 Hz on alternate minutes, depending on the exact time and to which station you are listening.

If you do NOT use the TCXO feature of the HS1:

For most users there is no reason **NOT** to use the TCXO feature of the HS1 as it will greatly improve frequency stability with changing temperature. The main reasons why one might not want to use it include:

- For testing/debugging. Perhaps, while building, the function does not work or the temperature sensor is not available.
- To characterize the temperature drift. If you wanted to modify the source code to model the Si570 in *your* transceiver to minimize drift you would turn it off and measure the frequency change at 14.000 MHz and apply these corrections.
- You are hearing a “tick-tick” sound in your audio – particularly on higher bands, caused by the data polling of the temperature data.

If you are having trouble with the “tick-tick” it may be that you are using a UI board that is older than revision 0.4 that has not been modified and/or you have not turned on the “**RX/TX Freq Xlate**” mode (*preferably to “**RX LO LOW**”*)

If, for some reason, the above steps do not work or you do not wish to perform them at this time, you may do the following:

- The calibration procedure is the same as above, but note that there will be a slight frequency shift when you switch between the TCXO ON and OFF (*or to the “Stopped”*) mode so be sure to perform the frequency calibration in whatever mode you intend to use.
- You should expect an observable and possibly significant frequency shift as the temperature changes!

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 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 "HS1 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 and waterfall display when “magnify” mode is turned off.
- **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 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 I2C communications of the temperature sensor*) 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 necessary for AM TX and for FM RX/TX operation.

Quirks and side-effects of the frequency translation mode:**Spectrum Scope/Waterfall Display offset:**

If the menu item “**Spec/Waterfall 2x 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!***

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.

Translation in transmit mode:

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.

Because of this frequency translation, in SSB transmit, you will also note that if you monitor the LINE OUT jack, you will no longer hear the SSB transmit audio directly. The reason for this is that there is only ONE D/A converter on the HS1 and with the frequency translation occurring, it is possible only to patch through the signal being fed to the modulator - which is no longer at "baseband." In theory, it should be possible to make a modification to the radio to use one of the existing 8-bit D/A channels to provide a "local" audio monitoring source, but this is something to be explored.

In CW mode things are a bit more complicated as there is the need for a sidetone - and the only way to generate a sidetone is via the monitoring of the audio being sent to the modulators. For this reason, frequency translation cannot be done in CW mode so the local oscillator must be shifted between receive and transmit - and **THIS** is where the bugs may show up again.

Effects in AM mode:

The use of frequency translation removes the problem of the “Hole” when using “AM” mode, eliminating the need to off-tune the signal to prevent its carrier from being at “zero IF”, in the middle. 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.

HS1 Key function matrix

Button(s)	Brief press	Press-and-hold >1 second
Power	Change display brightness	Power on and power off/save settings to memory
M1	Select AFG and CMP or STG	
M2	Select RFG and DSP or NB	Switch between DSP and NB
M3	Select RIT and MIC or LIN	Switch between MIC or LIN
G1	Change operational mode	Change operational mode - including disabled mode(s)
G2	Change DSP mode	Enable/Disable DSP without changing mode
G3	Change transmit power level	Generate CW Sidetone reference (CW/LSB/USB modes only)
G4	Change receive bandwidth (except in FM mode)	Change receive bandwidth – including disabled bandwidth(s) Generate tone burst (FM TX only)
BAND-	Change to different band	
BAND+	Change to different band	
STEP-	Change step size	Temporarily change to smaller tuning step size
STEP+	Change step size	Temporarily change to larger tuning step size
STEP- & STEP+	Lock/Unlock tuning knob	
POWER & BAND-	Toggle display backlight auto-off	
BAND- & BAND+	Toggle Spectrum Scope and Waterfall modes	

Main operational (receive/transmit) mode:

F1	Enter MENU mode	Save settings to memory
F2	Change meter mode	
F3	Toggle SPLIT mode on/off	
F4	Toggle VFO A/B	Copy active VFO to inactive VFO (A=B or B=A)
F5	Toggle TUNE mode	Enable/Disable transmit toggle

Menu mode:

F1	Exit MENU mode	Save settings to memory
F2	Set selected menu item to default	
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	Toggle TUNE mode	Enable/Disable transmit toggle

- Pressing-and-holding **F1**, **F3** and **F5** simultaneously while powering up the radio will cause defaults to be loaded. The user must either disconnect power to retain the old settings *or* press-and-hold the **POWER** button to overwrite them with defaults. ***All configurations, adjustments, frequencies and mode settings will be lost!***
- Pressing-and-holding a button (*other than POWER*) while powering up the radio will cause it to enter a “button test” mode. If multiple keys are pressed, only the name of the button with the highest precedence will be displayed. In decreasing order, the precedence is: M2, G3, G2, BNDM, G4, M3, STEPM, STEPP, M1, M3, F1, F2, F4, BNDP, F5, G1 and **POWER**
- The “**Band-**” and “**Band+**” and the “**Step-**” and “**Step+**” functions may be swapped using menu selections.

Operational notes, quirks, and known bugs:

- **CW operation will be impaired while in the menu system!** It is recommended that one **NOT** transmit in CW while the menu system is being displayed as the dit-dah timing will be disrupted!
- In the sudden presence of a strong, steady signal, a “Tick-tick” sound may be heard for a few seconds, accompanied by the lower portion (“S0-S9”) of the S-Meter turning red. This sound is the result of the automatic gain adjustment of the A/D converter in the codec being quickly adjusted to prevent overload, with the “tick” sounds being due to the large step sizes of the gain reduction. In normal operation, in the presence of modulation (*e.g. audio*) rather than a steady carrier this artifact is inaudible. *For more information see the menu item “RX Codec Gain”.*
- When the transceiver is powered up one of the pieces of information that is displayed is the interface mode of the LCD. The desired mode is “**Parallel**”, which is obtained with UI board revisions 0.3 and later with appropriately-jumpered HY-28B LCD displays. If you have have a version 0.3 board and this newer display, but the startup message indicates that the LCD interface mode is **SPI**, you may consider **CAREFULLY** removing the LCD and properly setting its jumpers to achieve parallel mode.
- If you have an older LCD and/or board that runs in SPI mode you may wish to change the parameter “**Spec. Scope Filter**” to 1 or 2 to reduce its strength and speed the response to changing signals.
- If you experience a one-second “Tick” sound on higher bands (15, 12, 10 meters) you may wish to perform the modification that suppresses this problems. Prior to making these modifications, this “tick” sound may be suppressed by setting the menu parameter “**TCXO Off/On/Stop**” to “Stop”. Note that this will halt polling of the temperature sensor, disabling the temperature display and the ability of the temperature drift of the synthesizer to be compensated for temperature change.

These modifications may be found in the “KA7OEI” folder on the Yahoo

group. Please make sure that the “Frequency Translate” function is active as this will also reduce this noise.

- There *may* be a lingering bug in the CW mode in which the transceiver momentarily “hangs” on rare occasions, particularly when going quickly from TX, to RX and then back to TX again. It is believed that this bug has been fixed, but if it does occur, increase the length of the “**CW TX- >RX Delay**” parameter slightly and please report its occurrence and the relating circumstances on the Yahoo group.
- If the various modifications to improve receiver performance have been performed (*e.g. the “U3a” mod, a separate regulator for the MCU, the 4.7 ohm resistor in the 8 volt supply for the audio amplifier, the resistor/filter for the LCD supply, etc.*) the receiver sensitivity will increase to the point that EMI from the LCD's data bus can get into the receiver. When the spectrum scope display is updated, this can cause a “Helicopter”-like sound which may be significantly reduced by placing a metal shield between the UI and RF boards. This shield may be any type of metal, but it must be insulated on both sides to prevent its shorting components. *This shield need not be grounded to effect significant improvement in receiver performance.* This effect is less apparent with the LCD operating in SPI mode. *The board modifications were incorporated into UI board versions ≥ 0.4 , but it is recommended that a metal shield be placed between the two boards no matter which version of boards you use.*
- Starting with version 0.0.211, the “build” number (“211” for example) is stored in the EEPROM and compared upon boot-up. If this is different, it is assumed that a new version of firmware has been loaded and new EEPROM variables are automatically initialized. Note that this is triggered **ONLY** if the build number of the loaded firmware is different from what was previously loaded into the radio.

- Please refer to the “board modification” document for information on reducing the amount of induced noise caused by the LCD when the display is dimmed. This document may be found in the FILES section of the YAHOO group in the KA7OEI folder. Changes have been incorporated in version 0.4 of the UI board to reduce this along with using the “Frequency Translate” function.
- When the LCD display is dimmed, it is a known that the display will flash somewhat when operating CW. It is hoped that this will be resolved in a later version of firmware.
- Starting with version 0.0.219.x the usage of “major” and “minor” version numbering has changed. Prior to this the “major” version numbering was not used (e.g. 0.0.0.211, 0.0.0.112). Now, both major and minor version numbering is used, as in “0.0.219.15”, “0.0.219.16”, etc.

Circuit description of the HS1:

The HS1 transceiver consists of two boards: The UI (User Interface) board that contains the MCU (computer) and display, along with the audio input/output and buttons and the RF (Radio Frequency) board that contains the power supply, frequency synthesizer, transmit and receive mixers, transmit power amplifiers and bandpass and lowpass filters. These two boards are connected together via a 30 pin SIP (Single Inline Pin) connector to form one compact unit.

RF Board:

Power Supply:

Located on the RF board, DC power, ranging from 9 to 16 volts, is input via J1, passing through fuse F1 which could be a one-time fuse or a self-resetting “PTC” type fuse, depending on the builder's choice. Following this is D1 that, in the event of inadvertent application of reverse polarity, will conduct and cause the fuse to blow, protecting the radio from permanent damage.

Providing a low impedance DC source for transceiver, C27 filters the input voltage upon application to the “unswitched” DC circuits (VCC_12V) and switchable regulator U3 while R13 and R14 scale the input voltage to a range suitable for measurement by the MCU with C31 providing the necessary low impedance AC input for its A/D input.

R9 and R10 form a voltage divider connected to the ON/OFF pin (pin 2) of voltage regulator U3 that, when pulled to ground, enables its output, turning on its output which is “programmed” for a nominal 8 volts via resistors R11

and R12. Pin 2 may be pulled down via diode pack D2 by either a press of the POWER button on the UI board or by an output of the MCU itself. In normal operation, the MCU keeps this pin ("PowerDown") low to keep the transceiver powered up, but releases it (e.g. allows it to go high) when a power down is permitted. This is why, when powering off the transceiver, it does not actually turn off until one releases the POWER button.

The output of U3, "VCC_8V" is provided to a number of places: To the LM386 audio amplifier on the UI board and to U4, a 5 volt regulator on the RF board. The output of the 5 volt regulator, "VCC_5V", is then, in turn, distributed to the majority of the circuits on the RF board as well as the main power for the LCD on the UI board in addition. The 5 volt supply is also used to power U5, a low-dropout 3.3 volt regulator on the RF board which is used to power the RF synthesizer and related circuitry and some of the audio circuitry on the UI board as well as a separate 3.3 volt low-dropout regulator present on the UI board used to power the MCU.

RF Synthesizer:

An Si570 clock generator (U8) is used as the RF signal source for the HS1 transceiver. This device, capable of part-per-billion tuning resolution, operates at four times the TX/RX frequency and is capable of tuning from at least 10 MHz (*2.5 MHz*) through 120 MHz (*30 MHz*) – although individual devices can typically tune beyond this range allowing, at least partial or complete coverage of the 160 meter amateur band. The Si570 is tuned via an SPI (serial) interface, controlled by the MCU and its RF output is first buffered by U9 to "square" it as the Si570's output could be a lower-level sine. The output of this is then fed to U11 which generates a quadrature signal at the TX/RX frequency. These signals are then fed to U12 and U13 which provide separate differential drive signals at 5 volt logic levels for the receive and transmit mixers.

U10 is an SPI-interface temperature sensor that is thermally bonded to U8, typically by using epoxy to adhere a piece of aluminum or copper to bridge the top of the two parts so that they track each other. The Si570, being originally intended for a generalized clock source in networking applications does not have a particularly stable internal crystal reference and as such, its absolute frequency can vary significantly with temperature. Because the nature of this variance is repeatable and dependent on temperature, the resulting frequency drift may be compensated by monitoring the Si570's temperature and applying corrections in software.

Antenna Switching and Lowpass Filtering:

The RF signals are applied via J1, a BNC connector. DS1, a standard neon discharge tube, with its 60-90 volt "breakover" voltage, offers a degree of protection to the transceiver against brief transients such as those related to lightning discharges while the DC continuity of T2/T3 prevent the accumulation of static charge.

Transformers T2 and T3 form what is known as a "Tandem Coupler", being directionally sensitive to the flow of RF current through the primary of T2. For example, if the load connected to J1 is matched to 50 ohms, a sample of that RF output will appear at the junction of R57 and D5 but none will appear at the junction of R61 and D6. If there is a mismatched load at J1, the amount of reflected power will be proportionally indicated by the amount of RF appearing at the junctions of R57/D5 and R61/D6 with the latter indicating the amount of reflected RF.

The signal feeding the Tandem Coupler "ANT_MET" comes from the lowpass filter selection which is accomplished through the use of magnetically latching relays. These have the advantage of needing to be

energized only briefly, holding their state indefinitely until change and thus saving power. By setting the appropriate combination of relays and contacts the appropriate low-pass filter may be placed inline. The relays themselves are driven by U14 which decodes a brief pulse from the MCU to place them in the desired configuration upon power-up and/or when the frequency/bands are changed.

The “input” side of the low-pass filtering is the “TX_PA_OUT” line which connects to two places:

The “top” of T7, the output matching filter for the power amplifier and the PIN diode T/R switch. When in receive mode PIN diode D4 is turned off and PIN diode D3 is turned on by setting lines “ANT_TX_ON” low and “ANT_RX_ON” high, respectively, providing a signal path from “TX_PA_OUT” to “RX_ANT” and to the receive section.

When in transmit mode, the “ANT_TX_ON” line goes high and “ANT_RX_ON” line goes low, causing diode D4 to turn on which shorts the signals at that point to ground and putting a reverse bias across D3, increasing its “off” isolation even more, further improving the isolation of the receiver input circuitry from the transmitter output.

It is worth noting that the PIN diodes in this circuit do not carry RF power as they are not in the transmit path, but must be rated to stand off the peak RF voltage that could occur at full RF output in a severe mismatch condition: The 200 volt rating of the specified diodes offers comfortable margin.

It is worth noting that in receive mode, the signal path is connected in parallel with RF output transformer, T7 and thus the RF finds themselves.

While this can cause a small amount of signal reduction owing to the losses of T7 and the shunting effects of the RF power transistors Q5 and Q6, this effect is rather minimal. This approach – quite common in QRP rigs – was taken to simplify the circuitry and eliminate the need to insert another low-loss switch, either a relay or high-power PIN diode, in the transmit signal path.

Bandpass Filters:

The bandpass filter network is used for both receive and transmit, the roles being selected by the configuration of switches U1 and U2 which both select which filter is used and which is connected to what input/output.

In receive mode the “RX_ANT” line from the PIN diode switch is routed to the appropriate filter, selected by control signals from the MCU via U1. Passing through the appropriate filter, the signal then passes through U2 which uses the same signals as U1 which then applies the now-filtered signal to the “RX_QSD_IN” line.

In transmit mode the PTT line switches U1 and U2 to the “other” set of switches, routing the signal from transmit mixer “TX_MIX”, through the filter, to the PA driver stage via “TX_PA_IN”.

Shown as “optional”, there is a resistor network that may be populated to provide a bias voltage for switches U1 and U2 in the bandpass filter network. At high signal levels biasing these switches somewhere around “mid supply” may improve linearity (reduce distortion) but this may not be noted under normal operation. If in doubt, these components may be installed with absolutely no deleterious side effects.

RX Detector:

The “RX_QSD_IN” signal from the bandpass filter is applied to RF preamplifier Q1 which has approximately 22 dB gain. This signal is then passed through T1 which produces a differential signal that is applied to a Quadrature Sampling Detector (a.k.a. “Tayloe” Detector).

This mixer works by turning on a switch at the desired receive frequency (e.g. the local oscillator frequency) for a portion of the RF cycle – and then turning it off again. If there was signal on the input of the detector was “close” to that of the local oscillator signal, some of its energy will be stored in the capacitor connected to its output and if the voltage on that capacitor is sensed and amplified, that signal can be detected as receive audio.

Because a single detector cannot adequately distinguish the sum and difference mixing frequencies, quadrature local oscillators (e.g. those 90 degrees apart at RF) are used to produce a pair of audio signals, “I” (in-phase) and “Q” (quadrature) which may then be used later on to distinguish between the sum and difference signals via the “phasing method” via mathematical methods in the MCU.

The function of the “switch” in the above description is provided by U15 with the “storage capacitors” being C68 and C69. Low noise op amp U16 provides both gain low-pass filtering to the signals from the QSD before being sent to the UI board for processing.

TX Quad Preamp and TX Mixer:

For transmitting, the I and Q signals from the UI board are filtered and buffered by U19 which also provides a set of 180 degree (differential)

signals for both I and Q. The two outputs from each channel are further-boosted to a low-impedance source using LM386 audio amplifiers, U20-U23.

The high-level, low-impedance differential audio signals are applied to U17, a switch which like U15 in the receive QSD, is driven at the operating (transmit) frequency – but in “reverse”. Instead of RF input producing audio on the output of the switch, audio is applied to the switch and double-sideband audio appears on the RF side of the each switch.

Were it not for the fact that both our local oscillator *and* our audio were produced in quadrature already, we would end up with double-sideband signals on the output of our mixer, but due to the math involved in the “phasing method” the undesired signals cancel out in the mixer, yielding only a single set of signals on the output of T4, the “TX_MIX” line which is then passed to the Bandpass Filter.

TX Power Amplifier:

The transmit signals from the mixer, having already passed through the bandpass filter, is first applied to T5 to produce a differential signal. With their drive signals 180 degrees apart, Q3 and Q4 each amplify the input signal, boosting its voltage significantly from a few hundred millivolts on the input to several volts on their respective collectors.

Note: In the original diagram the bias is applied continuously to Q3 and Q4 causing a continuous collector current of 50-70mA to flow, even when the transceiver is off. In the “Board Modifications” document there is information on a modification to change this to a “keyed” bias so that Q3 and Q4 are only biased when the PTT line is active.

This signal – at a fairly high impedance – is then applied, via DC blocking capacitors C99 and C100, to the final RF amplifier transistors Q5 and Q6, N-channel RF power FETs. The drain voltage for these FETs is provided by T6 which is bifilar wound with the winding phased such that RF “ground” (DC input) of one of the FET's connections is physically the same as the RF “hot” side of the other. RF output is extracted via T7, an RF transformer with a 2:3 impedance transformation with the “low” side of the impedance being connected to Q5 and Q6.

DC bias for the FETs is supplied via U18 which, when PTT is active, with the voltage being set via a D/A output from the MCU from the line “PA_BIAS”. This line varies from 0-3.3 volts, but the intrinsic 1.25 volt offset of U18 means that the actual voltage appearing on its pin 1 may be varied from 1.25 to 4.55 volts, nominal.

Notes:

- It is necessary that an additional resistor of between 1k and 10k be placed across C96 in order to establish a minimum DC current load for U18. Because FETs draw no gate current, it is possible that normal device leakage current of the FET and/or U18 can cause this voltage to exceed the set voltage if there is no current load present.
- In addition to the parallel resistance mentioned above, additional capacitance is **required** in parallel with C96 to assure that U18 operates in a stable mode and does not oscillate, causing distorted transmit signals. **A minimum of a 22 uF Tantalum (or 100 uF electrolytic) is what is needed to assure stability of this circuit under all operating conditions!**
- When ordering final FETs for the HS1, it is recommended that one order “matching” FETs if possible. If the option of “matched” devices is not available, it is recommended that “extra” devices be obtained

and then two devices selected for closest-matching threshold voltages. Because there is only one bias setting, these FETs should be “matched” in terms of DC threshold voltage. This voltage may be easily determined with the use of one of those <\$20US universal “component” testers, or a simple test circuit constructed to ascertain this threshold.

UI Board:

MCU power and clocks:

The heart of the HS1 transceiver is the MCU, an STM32F405-VG or STM32F407-VG (*either one will work*), a processor with a Cortex ARM M4 with hardware floating point, internally clocked at 168 MHz with 1 Megabyte of flash program memory and 192 kbytes of SRAM. Using **either** a 20 MHz TCXO (U5) **or** a 20 MHz crystal (Y1) (*builder's option*) the processor's clock, and the clock for the audio codec are synthesized: This clock source has nothing to do with the main RF synthesizer frequency or its accuracy except by the accuracy of the sample rate of the audio codec.

The MCU has its own power source, derived from the +5 volt supply (UI_5V) from U6, isolated via R45, R46 and C96. This added regulator and isolation were found to be necessary as the MCU's current requirements caused a very slight modulation of the main 3.3 volt rail in the original design which resulted in circulating currents in the ground of the HS1 and thus appearing at the millivolt level (or lower) in the receive I and Q audio lines: It was only by adding this extra regulator and the R/C filtering and isolating this “noise” to the physical vicinity of the MCU that his noise source was eliminated.

The presence of jumper P6 should be noted. It is used only upon the initial

installation of the HS1 bootloader, replacing the factory-installed bootloader. After this has been successfully completed, P6 should be removed and not needed again.

On version 0.4 of the RF board U7, an accommodation to allow the addition a serial EEPROM has been provided. As of this firmware version, it is not supported and it need not be installed.

Front Panel User Controls and RF Board Interface and USB:

The main user interface consists of four rotary encoders and seventeen pushbuttons, all interfaced directly to the MCU's IO lines utilizing onboard pull-ups. Each of these lines has included a bypass capacitor, both for debounce purposes and also to provide a degree of RF immunity as well as to reduce the probability of spurious RF emissions from those lines.

Also controlled by the interface are two LEDs. The Green LED indicates that the MCU is operational and powered up while the Red LED is used to show that the PTT (transmit) is active.

For the newer boards (Version 0.3 and up) the typical LCD used, the HY28B, has a parallel interface that allows for the direct mapping of the display's RAM (256kbytes) to the memory management peripheral of the MCU allowing objects to be drawn on the screen simply by writing to the appropriate memory location.

For older boards that use the HY28A, the interface to the LCD is based on the SPI (serial) interface which is very much slower, but a separate set of optimized display routines have been implemented in the HS1's code to maximize speed and usability.

Pin 40 of the LCD ("BL_CTRL") is used to blank the backlight and it is also PWM (Pulse-Width Modulated) controlled to provide a brightness control of the LCD to save power. Because the use of PWM necessarily modulates the LCD's power supply, additional filtering in the form of C74a and R35a have been added to keep this modulation energy from finding its way onto the main "UI_5V" bus, onto the ground bus and into the low-level audio. At the present time the backlight dimming PWM signal is generated in software and there are some operational conditions (*e.g. when transmitting using CW*) that the brightness can vary due to disruptions in this waveform.

The "RF Board Interface and USB" provides the pathways from the MCU to the RF board. Of note are the SPI signals "SCL" and "SDA" and resistors R47 and R48: These resistors reduce the slew rate of the falling edge of the serial data from the MCU to the Si570 and the temperature sensor, minimizing the generation of broadband RF energy and a period "tick" that may result.

Also included is J11, the USB Dfu connector used for programming and J10, the full-sized USB connector which is reserved for future use with other peripherals. Connector P8 is also provided, but its use is limited for hardware debug using the appropriate development tools.

Codec and Audio Switching:

The receive I and Q audio from U16 on the RF board is fed into U3 which is used to select either the receive mixer or the "Line In" jack as there is only one A/D converter on the HS1. In receive mode, U3 always routes the audio from U16 to the "LLINEIN" and "RLINEIN" pins of U1, the codec where the audio is digitized and made available to the MCU via a dedicated SPI interface.

After being processed by the MCU, the receive audio is sent back to U1 via the SPI interface and it is output on the output pins. On this particular codec, there are two sets of outputs: “LHPOUT” and “RHPOUT” which have adjustable output levels and “LOUT” and “ROUT” which have fixed levels – but they both carry the exact same audio.

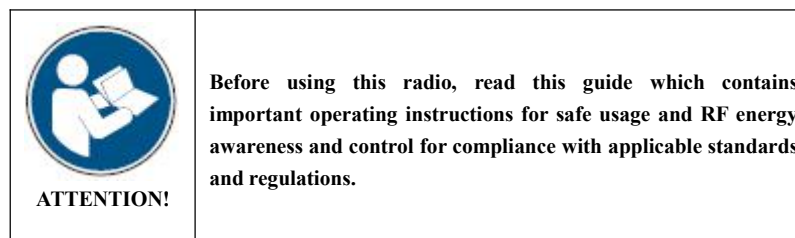
Speaker audio is output via the “LHPOUT” line to U2, an LM386 audio amplifier while the “Line Out” audio is output from the “RHPOUT” line. Because they are independently adjustable, their levels may be varied and muted as needed – via the volume control in the case of the speaker for receive and muted during transmit for both the speaker and Line Out.

In transmit mode, there are two possible sources of transmit audio: If the “Line In” mode is selected the codec is configured to use audio from its two “Line In” pins as U3 always routes the signal from the “Line In” jack to these pins when in transmit mode. If the “Microphone Input” is selected, the codec uses only the “MICIN” pin for its audio source. The selected audio is then digitized and sent to the MCU to be processed.

The processed audio to be transmitted is then sent back to the codec to be converted back to analog and it is output on the “LOUT” and “ROUT” pins. When in transmit mode analog switch U3a is closed, allowing the signal from these two pins to be sent to the RF board in the “AUDIO_OUT_I” and “AUDIO_OUT_Q” lines.

On UI boards 0.3 and earlier, switch U3a was absent, which posed a problem when in receive mode. Because the receive audio is always present on the “LOUT” and “ROUT” pins, receive audio was also sent on the “AUDIO_OUT_I” and “AUDIO_OUT_Q” lines to the RF board, amplified, and applied to the transmit QSD mixer, U17. At high receive

volumes and with strong signals the audio going into U17 exceeded the 5 volt supply voltage of that device causing it to falsely turn on, creating onfrequency spurious signals which, in turn, resulted in distorted feedback in the receiver. The addition of U3a permits the blocking of these signals when in receive mode, eliminating this problem entirely.

RF ENERGY EXPOSURE AND PRODUCT SAFETY GUIDE FOR PORTABLE TWO-WAY RADIOS

This two-way radio uses electromagnetic energy in the radio frequency (RF) spectrum to provide communications between two or more users over a distance. RF energy, which when used improperly, can cause biological damage.

All Retevis two-way radios are designed, manufactured, and tested to ensure they meet government-established RF exposure levels. In addition, manufacturers also recommend specific operating instructions to users of two-way radios. These instructions are important because they inform users about RF energy exposure and provide simple procedures on how to control it.

Please refer to the following websites for more information on what RF energy exposure is and how to control your exposure to assure compliance with established RF exposure limits: <http://www.who.int/en/>

Local Government Regulations

When two-way radios are used as a consequence of employment, the Local Government Regulations requires users to be fully aware of and able to control their exposure to meet occupational requirements. Exposure awareness can be facilitated by the use of a product label directing users to specific user awareness information. Your Retevis two-way radio has a RF Exposure Product Label. Also, your Retevis user manual, or separate safety booklet includes information and operating instructions required to control your RF exposure and to satisfy compliance requirements.

Radio License

Governments keep the radios in classification, business two-way radios operate on radio frequencies that are regulated by the local radio management departments (FCC, ISED, OFCOM, ANFR, BFTK, Bundesnetzagentur...). To transmit on these frequencies, you are required to have a license issued by them. The detailed classification and the use of your two radios, please contact the local government radio management departments.

Use of this radio outside the country where it was intended to be distributed is subject to government regulations and may be prohibited.

Unauthorized modification and adjustment

Changes or modifications not expressly approved by the party responsible for compliance may void the user's authority granted by the local government radio management departments to operate this radio and should not be made. To comply with the corresponding requirements, transmitter adjustments should be made only by or under the supervision of a person certified as technically qualified to perform transmitter maintenance and repairs in the private land mobile and fixed services as certified by an organization representative of the user of those services.

Replacement of any transmitter component (crystal, semiconductor, etc.) not authorized by the local government radio management departments equipment authorization for this radio could violate the rules.

FCC Requirements:

This device complies with part 15 of the FCC Rules. Operation is subject to the condition that this device does not cause harmful interference. (Licensed radios are applicable);

This device complies with part 15 of the FCC Rules. Operation is subject to the following two conditions: (Other devices are applicable)

- (1) This device may not cause harmful interference, and
- (2) this device must accept any interference received, including interference that may cause undesired operation.

NOTE: This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

CE Requirements:

•(Simple EU declaration of conformity) Shenzhen Retevis Technology Co., Ltd. declares that the radio equipment type is in compliance with the essential requirements and other relevant provisions of RED Directive 2014/53/EU and the ROHS Directive 2011/65/EU and the WEEE Directive 2012/19/EU; the full text of the EU declaration of conformity is available at the following internet address: www.retevis.com.

•Restriction Information

This product can be used in EU countries and regions, including: Belgium (BE), Bulgaria (BG), Czech Republic (CZ), Denmark (DK), Germany (DE), Estonia (EE), Ireland (IE), Greece (EL), Spain (ES), France (FR), Croatia (HR), Italy (IT), Cyprus (CY), Latvia (LV), Lithuania (LT), Luxembourg (LU), Hungary (HU), Malta (MT), Netherlands (NL), Austria (AT), Poland (PL), Portugal (PT), Romania (RO), Slovenia (SI), Slovakia (SK), Finland (FI), Sweden (SE) and United Kingdom (UK).

For the warning information of the frequency restriction, please refer to the package or manual section.

•Disposal

The crossed-out wheeled-bin symbol on your product, literature, or packaging reminds you that in the European Union, all electrical and electronic products, batteries, and accumulators (rechargeable batteries) must be taken to designated collection locations at the end of their working life. Do not dispose of these products as unsorted municipal waste. Dispose of them according to the laws in your area. ■



IC Requirements:

Licence-exempt radio apparatus

This device contains licence-exempt transmitter(s)/receiver(s) that comply with Innovation, Science and Economic Development Canada's licence-exempt RSS(s). Operation is subject to the following two conditions:

- (1) This device may not cause interference.
- (2) This device must accept any interference, including interference that may cause undesired operation of the device.

Le présent appareil est conforme aux CNR d'Industrie Canada applicables aux appareils radio exempts de licence. L'exploitation est autorisée aux deux conditions suivantes :

- (1) l'appareil ne doit pas produire de brouillage;
- (2) l'utilisateur de l'appareil doit accepter tout brouillage radioélectrique subi, même si le brouillage est susceptible d'en compromettre le fonctionnement.

RF Exposure Information

- DO NOT operate the radio without a proper antenna attached, as this may damage the radio and may also cause you to exceed RF exposure limits. A proper antenna is the antenna supplied with this radio by the manufacturer or an antenna specifically authorized by the manufacturer for use with this radio, and the antenna gain shall not exceed the specified gain by the manufacturer declared.
- DO NOT transmit for more than 50% of total radio use time, more than 50% of the time can cause RF exposure compliance requirements to be exceeded.
- During transmissions, your radio generates RF energy that can possibly cause interference with other devices or systems. To avoid such interference, turn off the radio in areas where signs are posted to do so.
- DO NOT operate the transmitter in areas that are sensitive to electromagnetic radiation such as hospitals, aircraft, and blasting sites.
- Portable Device, this transmitter may operate with the antenna(s) documented in this filing in Push-to-Talk and body-worn configurations. RF exposure compliance is limited to the specific belt-clip and accessory configurations as documented in this filing and the separation distance between user and the device or its antenna shall be at least 2.5 cm.
- Mobile Device, during operation, the separation distance between user and the antenna subjects to actual regulations, this separation distance will ensure that there is sufficient distance from a properly installed

externally-mounted antenna to satisfy the RF exposure requirements.

- Occupational/Controlled Radio, this radio is designed for and classified as “Occupational/Controlled Use Only”, meaning it must be used only during the course of employment by individuals aware of the hazards, and the ways to minimize such hazards; NOT intended for use in a General population/uncontrolled environment.
- General population/uncontrolled Radio, this radio is designed for and classified as “General population/uncontrolled Use”.

RF Exposure Compliance and Control Guidelines and Operating Instructions

To control your exposure and ensure compliance with the occupational/controlled environment exposure limits, always adhere to the following procedures.

Guidelines:

- User awareness instructions should accompany the device when transferred to other users.
- Do not use this device if the operational requirements described herein are not met.

Operating Instructions:

- Transmit no more than the rated duty factor of 50% of the time. To Transmit (Talk), push the Push to Talk (PTT) button. To receive calls (listen), release the PTT button. Transmitting 50% of the time, or less, is important because the radio generates measurable RF energy exposure only when transmitting in terms of measuring for standards compliance.
- Transmit only when people outside the vehicle are at least the recommended minimum lateral distance away from a properly installed according to installation instructions, externally mounted antenna.
- When operating in front of the face, worn on the body, always place the radio in a Retevis approved clip, holder, holster, case, or body harness for this product. Using approved body-worn accessories is important because the use of Non-Retevis approved accessories may result in exposure levels, which exceed the IEEE/ICNIRP RF exposure limits.

Hand-held Mode

- Hold the radio in a vertical position with the microphone (and other parts of the radio including the antenna) at least 2.5 cm (one inch) away from the nose or lips. The antenna should be kept away from the eyes. Keeping the radio at a proper distance is important as RF exposure decreases with increasing distance from the antenna.



Phone Mode

- When placing or receiving a phone call, hold your radio product as you would a wireless telephone. Speak directly into the microphone.

Electromagnetic Interference/Compatibility

NOTE: Nearly every electronic device is susceptible to electromagnetic interference (EMI) if inadequately shielded, designed, or otherwise configured for electromagnetic compatibility.

Avoid Choking Hazard



Small Parts. Not for children under 3 years.

Turn off your radio power in the following conditions:

**WARNING**

- Turn off your radio before removing (installing) a battery or accessory or when charging battery.
 - Turn off your radio when you are in a potentially hazardous environments: Near electrical blasting caps, in a blasting area, in explosive atmospheres (flammable gas, dust particles, metallic powders, grain powders, etc.).
 - Turn off your radio while taking on fuel or while parked at gasoline service stations.
- To avoid electromagnetic interference and/or compatibility conflicts
- Turn off your radio in any facility where posted notices instruct you to do so, hospitals or health care facilities (Pacemakers, Hearing Aids and Other Medical Devices) may be using equipment that is sensitive to external RF energy.
 - Turn off your radio when on board an aircraft. Any use of a radio must be in accordance with applicable regulations per airline crew instructions.

Protect your hearing:**WARNING**

- Use the lowest volume necessary to do your job.
 - Turn up the volume only if you are in noisy surroundings.
 - Turn down the volume before adding headset or earpiece.
 - Limit the amount of time you use headsets or earpieces at high volume.
 - When using the radio without a headset or earpiece, do not place the radio's speaker directly against your ear
 - Use careful with the earphone maybe possible excessive sound pressure from earphones and headphones can
- Note:** Exposure to loud noises from any source for extended periods of time may temporarily or permanently the less time is required before your hearing could be affected. Hearing damage from loud noise is sometimes



cause hearing loss affect your hearing. The louder the radio's volume, undetectable at first and can have a cumulative effect.

Avoid Burns**WARNING****Antennas**

- Do not use any portable radio that has a damaged antenna. If a damaged antenna comes into contact with the skin when the radio is in use, a minor burn can result.

Batteries (If appropriate)

- When the conductive material such as jewelry, keys or chains touch exposed terminals of the batteries, may complete an electrical circuit (short circuit the battery) and become hot to cause bodily injury such as burns. Exercise care in handling any battery, particularly when placing it inside a pocket, purse or other container with metal objects

Long transmission

- When the transceiver is used for long transmissions, the radiator and chassis will become hot.

Safety Operation

**WARNING****Forbid**

- Do not use charger outdoors or in moist environments, use only in dry locations/conditions.
- Do not disassemble the charger, that may result in risk of electrical shock or fire.
- Do not operate the charger if it has been broken or damaged in any way.
- Do not place a portable radio in the area over an air bag or in the air bag deployment area. The radio may be propelled with great force and cause serious injury to occupants of the vehicle when the air bag inflates.

To reduce risk

- Pull by the plug rather than the cord when disconnecting the charger.
- Unplug the charger from the AC outlet before attempting any maintenance or cleaning.
- Contact Retevis for assistance regarding repairs and service.
- The adapter shall be installed near the equipment and shall be easily accessible

Approved Accessories**WARNING**

- This radio meets the RF exposure guidelines when used with the Retevis accessories supplied or designated for the product. Use of other accessories may not ensure compliance with the RF exposure guidelines and may violate regulations.
- For a list of Retevis-approved accessories for your radio model, visit the following website: <http://www.Retevis.com>