Version 0.0.217

February 16, 2015

Changes from version 0.0.215 include:

- **Added ability to press-and-hold of button G1 to force the selection of disabled modes.**
- **Added ability to press-and-hold button G4 to force the selection of disabled bandwidths.**
- **Added function to automatically select the sideband (LSB/USB) for the current band.**
- **Added a "Frequency Translate" function to the receiver *(and transmitter)* to reduce the effects of "DC Conversion" on the receiver.**
- **Normal AM receive mode is now possible.**
- **AM Transmit mode is now supported.**

IMPORTANT:

> **For unknown reasons the DSP filtering will occasionally "crash" but this version includes modifications to minimize the likelihood of this happening.**
>
> **If the DSP does crash, it is likely that this will be detected fairly quickly and the DSP reset automatically. If the automatic DSP reset does not occur, turning DSP off and back on will now reset it.**

Details:

Added ability to press-and-hold of button G1 to force the selection of disabled modes:

By pressing-and-holding button **G1** (the "Mode" button) a mode disabled in the menu may be enabled. As of software version 0.0.217, the only mode that can be disabled in software is AM.

This allows one to select AM for general coverage receiver reception, but still keep it out of the normal rotation when changing modes.

This feature allows the "opposite" sideband to be selected when the "**LSB/USB Auto Select**" mode is enabled – ***see below***.

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.

Added ability to press-and-hold button G4 to force the selection of disabled bandwidths:

By pressing-and-holding button **G4** (the "Bandwidth" button) a filter bandwidth disabled in the menu may be selected. This means that even if you have set menu item "**10k Filter**" to **OFF** or "**CW Filt in SSB Mode**" to **OFF**, you may still select the 10 kHz filter of the 300 Hz/500 Hz filters in any mode, respectively.

Added function to automatically select the sideband for the current band:

The menu item “**LSB/USB Auto Select**” has been added to automatically select LSB or USB depending on the current band.

There are three possible settings:

- **OFF** – No automatic selection.
- **ON** – LSB is selected < 10 MHz, USB is selected >= 10 MHz
- **USB 60M** – LSB is selected < 10 MHz *except* for 60 meters and USB is selected >= 10 MHz. This setting has been provided for those areas where USB is typically used on 60 meters *(e.g. the U.S.)*

Note: As noted above, pressing-and-holding button **G1** will allow one to select modes that would not normally be available. In this case, pressing-and-holding this button would allow the selection of LSB on 20 meters, for example. *Note that in order to select USB on 75 meters, you must first go through several other modes first.*

Added a “Frequency Translate” function to the receiver *(and transmitter)* to reduce the effects of “DC Conversion” on the receiver:

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 "mcHF 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!**

Additional options may be added in the future.

Quirks and side-effects:

When the translate mode is activated, 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!***

As it turns out, I have applied this frequency translation on SSB transmit as well. This should slightly improve the SSB audio quality and it also makes it ***theoretically*** possible to implement the transmission of AM signals as well with single and/or dual sidebands.

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 mcHF 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 (be on the lookout, Alain!)

I have hopefully "pre-empted" the former issues that had plagued the doomed version 0.0.213 by implementing the frequency shifting in a more clever manner than I'd attempted with that version. While I have not spotted any problems in my 20-30 minutes of messing about in CW, it will likely take someone who has gotten a good "feel" of how the rig behaves to tell me if I've messed anything up - and in what way.

Normal AM reception now possible:

With the frequency translation activated, the DC "Hole" that is present when frequency translation is turned off is now eliminated. This means that one may tune AM signals "normally", not having to avoid this "hole" with the AM carrier.

This can only be done ***when frequency translation is enabled!***

AM Transmit mode is now supported:

With frequency translation activated, full-carrier, double-sideband AM transmission is now possible. You should remember several things about AM:

- It is **MUCH** less efficient than SSB! You will have 1/8th of the "talk" power of SSB *(that's 9 dB!)* – 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 under any conditions.

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".

Warning:

> **The presence of a continuous carrier when transmitting AM can add significant thermal stress to the final amplifier transistors and your power supply!**
>
> **YOU HAVE BEEN WARNED!**

Comment:

This version was compiled with GCC Version 4.9.x.x rather than 4.7.x.x. It should be noted that a few things will be "broken" upon compile *(e.g. the spectrum scope when in AM mode)* unless optimization "-O1" mode is selected. *(If version 4.7.x.x is used, "No Optimization" may be selected.)*

Version 0.0.215

February 8, 2015

Changes from version 0.0.213 include:

- **Added multiple frequency/display CW shift options**
- **Changed the DSP NR (Noise Reduction) strength setting**
- **Added more preventative measures to reduce the likelihood of the DSP NR crashing**
- **Added automatic and manual reset of the DSP NR in the event that it crashes**
- **Added option to swap Band-/Band+ buttons**
- **Additional "un-mute" of audio on start-up**
- **Fixed problem in which adjustment of the MIC gain from the main panel while in receive mode caused the receiver to go deaf**

IMPORTANT:

For unknown reasons the DSP filtering will occasionally "crash" but this version includes modifications to minimize the likelihood of this happening.

If the DSP does crash, it is likely that this will be detected fairly quickly and the DSP reset automatically. If the automatic DSP reset does not occur, turning DSP off and back on will now reset it.

Details:

Added multiple frequency/display CW shift options:

Amongst the various radio manufacturers, the way the display and frequency offset is handled when in CW is handled differently. The on-screen display of the CW mode now displays "**CW-U**" and "**CW-L**" to indicate that the transceiver is operating in USB or LSB mode, respectively.

There are now ***nine*** settings to allow the user to select the method of frequency display/offset that suits them when operating CW under the **CW Freq. Offset** menu item:

- **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.

Changed the DSP NR (Noise Reduction) strength setting:

The "strength" of the DSP Noise Reduction (NR) setting is now being calculated differently – and the range of adjustment has been increased from 0-35 to 0-55. A previous setting of 15-20 is now equivalent to a setting in the range of 10-15. The DSP setting will also change color as the strength increases, going from yellow to orange to red. These color changes should be considered as a warning to indicate that they may not be useful in all situations.

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 voice!

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 "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.

Added more preventative measures to reduce the likelihood of the DSP NR crashing:

Additional measures have been added to prevent the DSP Noise reduction algorithm from crashing. For the most part this amounts to disabling DSP during certain operations, such as power-up, during transmit, and during band changes. Doing this prevents the DSP from being exposed to strong impulses which appear to be the culprit in causing the adaptive algorithm which comprises the parameters of the DSP to diverge uncontrollably.

Added automatic and manual reset of the DSP NR in the event that it crashes:

The DSP can now be unconditionally reset by turning it off and then back on using button **G2** – either by pressing it to cycle through the settings or by pressing-and-holding the button to turn it off, and then doing so again to turn it back on.

Automatic detection of the DSP crashing has also been implemented. While this will detect the majority of instance where the DSP will crash – and then reset the DSP – it will not always do so, or at least do so quickly, particularly when the DSP NR set set to a very high value (e.g. orange/red).

Added option to swap Band-/Band+ buttons:

If, for some reason, you feel inclined to swap the positions of the **Band-** and **Band+** buttons *(perhaps because you also have enabled the option to swap the **Step-** and **Step+** buttons)* you may now do this using the configuration menu item **Band+/- Button Swap**.

Additional "un-mute" of audio on start-up:

For various reasons, the mcHF's audio is occasionally muted on power-up – a problem "fixed" by adjusting the volume control. Additional steps have been taken in this version to prevent this from happening.

Fixed problem in which adjustment of the MIC gain from the main panel while in receive mode caused the receiver to go deaf:

It was pointed out that adjusting the Microphone gain from the main panel (but not in the menu) while in receive mode caused the receiver to go deaf until one briefly went into TX mode to reset the hardware. Even though this problem had likely been around since the Microphone gain was made available there, it had not been reported until recently: This problem has now been fixed!

Version 0.0.213

January 24, 2015

NOTE: This version is based on version 0.0.209 rather than 0.0.211. Due to various problems, the branch from 0.0.211 was scrapped. Many – but not all – of the changes made in versions later than 0.0.209 are included.

It is recommended that one not use version 0.0.211, particularly if you use CW!

This version is being released mostly to produce a current, stable version rather than introduce any major, new features.

Changes from version 0.0.209 include:

- **Fixed problem where Spectrum Scope wasn't redrawn after EEPROM save.**
- **A "Sanity Check" is done with readings from the temperature sensor to prevent red digits under very cold (or very hot) temperatures.**
- **The "Off" function has been re-added to DSP button G2 in voice mode.**
- **The DSP is not enabled until 3.5 seconds after start-up, this to reduce the probability of it being "stuck" (not working) and having to re-power the radio.**
- **The update of firmware is automatically detected and new EEPROM variables automatically initialized.**
- **Available adjustment ranges and defaults of the DSP NOTCH configuration settings have been changed to more useful ranges and values.**
- **DSP is disabled during TX and re-enabled on RX.**
- **Front-panel switching of settings has been disabled for CW mode.**
- **Added adjustment of Noise Blanker AGC factor.**
- **Synthesizer is not re-tuned when RIT = 0.**
- **Frequency Calibration now uses main tuning dial step sizes.**

IMPORTANT:

> **For unknown reasons the DSP filtering will occasionally "crash." If you power up the mcHF and get either no audio or badly distorted audio when DSP is enabled, or if this occurs when you are listening, the only way currently known work-around is to disable DSP or power-cycle the transceiver. The cause/fix for this is currently being investigated.**
>
> **This version includes some modifications that should significantly reduce the likelihood of the DSP crashing during normal operation, particularly during power-up and when going between RX and TX, but it is still possible.**

Fixed problem where Spectrum Scope wasn't redrawn after EEPROM save:

No-one else mentioned this, but pressing and holding the **F1** button to save settings to EEPROM while in normal receive mode resulted in the Spectrum Scope grid and background not being completely redrawn.

A "Sanity Check" is done with readings from the temperature sensor to prevent red digits under very cold (or very hot) temperatures:

It had been reported that at very cold temperatures that there was a "Red Digit" problem. This was due to the fact that the temperature-to-compensation lookup table only goes from 0C to 100C and was outside this range, causing "bogus" values to be passed to the compensation function. Now, readings outside the 0C to 100C range will use "default" (e.g. un-compensated) values and while the radio will now function, its frequency will not be accurate until the sensor/readout shows a temperature within the 0-100C range.

If you operate in an environment where the temperature regularly exceeds 100C, I suggest that you go somewhere else!

The "Off" function has been re-added to DSP button G2 in voice mode:

The "Off" setting has been added to the available settings when one quickly presses button **G2**. You can still press-and-hold this button to turn off the DSP, and then press-and-hold it to turn it back on with the same setting, useful if you have "**NR+NOT**" enabled so that you do not have to press the button multiple times.

The DSP is not enabled until 3.5 seconds after start-up, this to reduce the probability of it being "stuck" (not working) and having to re-power the radio:

One of the ongoing issues with the DSP is that it will occasionally "crash" - particularly if it is enabled on power-up. Delaying enabling the DSP until the radio is fully-initialized and operating may reduce the problem that occurs when one turns on the radio, finds the audio "dead" in DSP mode, and then having to power-cycle. The rare problem of the DSP suddenly crashing during normal operation is still being investigated.

The update of firmware is automatically detected and new EEPROM variables automatically initialized:

Starting with this version, the "build" number ("211" in this version) is stored in the EEPROM and compared upon bootup. 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.

Available adjustment ranges and defaults of the DSP NOTCH configuration settings have been changed to more useful ranges and values:

A bit of experimentation has revealed that new default values for "**DSP Notch ConvRate**" and "**DSP Notch BufLen**" are better than the previous default values: These may be obtained by going to these settings and pressing the **F2** (DEFLT) button. The available ranges both of these settings has been changed to more reasonable values.

The RX DSP is disabled during TX and re-enabled on RX:

If you notice very carefully, you will note if you have the DSP turned on (NR or notch) it is disabled for a split second after the radio returns from TX mode. One of the apparent causes of the DSP crashing is the loud "crash" that occurs (but is generally muted – but those using much earlier versions may remember!) when going between TX and RX. This sort of strong impulse appears to be a contributing factor in "crashing" the DSP.

Disabling the DSP and thereby "freezing" its filter coefficients also preserves its state, preventing that "crash" from altering the nature of the DSP filter, allowing it to filter better, more quickly upon return from transmit as well.

Front-panel switching of settings has been disabled for CW mode:

In version 0.0.209 and 0.0.211, setting the menu parameter #203 "O/S Menu SW on TX" to ON caused the main display to switch between AFG to STG and RIT to WPM while in TX mode. This has been disabled in CW mode since it slowed down the operation of CW mode.

Added adjustment of Noise Blanker AGC factor:

Parameter “NB AGC T/C (<=Slow)” has been added to the configuration menu. 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.

Note that while the menu is enabled, the noise blanker is always disabled, so you must exit the menu to note the effect of that parameter!

Synthesizer is not re-tuned when RIT = 0:

In order to speed up the response of the transceiver while in CW mode, extra code/logic was added, removing an unneeded “re-tune” command when going between RX and TX if the RIT offset is set to zero. In CW, particularly at higher speeds, this may be observed as slightly improved response.

Work is ongoing to further-improve the response of the transceiver in CW mode.

Frequency Calibration now uses main tuning dial step sizes:

The “Freq. Calibrate” parameter in the “Adjustment Menu” now uses the same frequency step sizes as the main frequency dial to make large calibration adjustments much easier. The adjustment range has been increased to +/- 9999 Hz.

Comment:

- If the LCD backlight is set to a dim mode, it will flash briefly during CW operation. This is a known issue that will (hopefully) be addressed in a forthcoming firmware version.

Version 0.0.211

January 2, 2015

Changes include:

- **Fixed problem where Spectrum Scope wasn't redrawn after EEPROM save.**
- **A “Sanity Check” is done with readings from the temperature sensor to prevent red digits under very cold (or very hot) temperatures.**
- **The “Off” function has been re-added to DSP button G2 in voice mode.**
- **The DSP is not enabled until 2 seconds after start-up, this to reduce the probability of it being “stuck” (not working) and having to re-power the radio.**
- **The loss of communications to the temperature sensor and/or Si570 is automatically detected and the serial bus is reinitialized to attempt to correct.**
- **The clocking rate of the serial bus has been slowed, allowing “stronger” EMI filtering to further-reduce the one-second “tick” on higher bands.**
- **The update of firmware is automatically detected and new EEPROM variables automatically initialized.**
- **Available adjustment ranges and defaults of the DSP NOTCH configuration settings have been changed to more useful ranges and values.**

IMPORTANT:

For unknown reasons the DSP filtering will occasionally “crash.” If you power up the mcHF and get either no audio or badly distorted audio when DSP is enabled, or if this occurs when you are listening, the only way currently known work-around is to disable DSP or power-cycle the transceiver. The cause/fix for this is currently being investigated.

Details:

Fixed problem where Spectrum Scope wasn't redrawn after EEPROM save:

No-one else mentioned this, but pressing and holding the **F1** button to save settings to EEPROM while in normal receive mode resulted in the Spectrum Scope grid and background not being completely redrawn.

A “Sanity Check” is done with readings from the temperature sensor to prevent red digits under very cold (or very hot) temperatures:

It had been reported that at very cold temperatures that there was a “Red Digit” problem. This was due to the fact that the temperature-to-compensation lookup table only goes from 0C to 100C and was outside this range, causing “bogus” values to be passed to the compensation function. Now, readings outside the 0C to 100C range will use “default” (e.g. un-compensated) values and while the radio will now function, its frequency will not be accurate until the sensor/readout shows a temperature within the 0-100C range.

If you operate in an environment where the temperature regularly exceeds 100C, I suggest that you go somewhere else!

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The “Off” setting has been added to the available settings when one quickly presses button **G2**. You can still press-and-hold this button to turn off the DSP, and then press-and-hold it to turn it back on with the same setting, useful if you have “**NR+NOT**” enabled so that you do not have to press the button multiple times.

The DSP is not enabled until 2 seconds after start-up, this to reduce the probability of it being "stuck" (not working) and having to re-power the radio:

One of the ongoing issues with the DSP is that it will occasionally "crash" - particularly if it is enabled on power-up. Delaying enabling the DSP until the radio is fully-initialized and operating may reduce the problem that occurs when one turns on the radio, finds the audio "dead" in DSP mode, and then having to power-cycle. The rare problem of the DSP suddenly crashing during normal operation is still being investigated.

The loss of communications to the temperature sensor and/or Si570 is automatically detected and the serial bus is reinitialized to attempt to correct:

If, for some reason, the communications with the Si570 and/or temperature sensor is lost – something that would often cause red digits and "pausing" of the spectrum scope and the user interface – it was observed that the only way to restore communications was to power-cycle the unit.

Now, this loss of communications is detected and the serial bus to these devices is reset. When this happens, a message will appear along with an audible disruption as the Si570 is reset.

It is believed that this function will detect if the temperature sensor is NOT installed, but this is not known for absolute certainty! If you do not have the temperature sensor installed, go into the menu, set the "**TCXO Off/On/Stop**" setting to **STOP** – which should stop the automatic resetting due to the lack of the sensor – and let me know about this.

The clocking rate of the serial bus has been slowed, allowing "stronger" EMI filtering to further-reduce the one-second "tick" on higher bands:

The clocking rate on the serial bus communicating with the Si570 (U8) and the MCP9081 (U10) has been lowered from 100 kHz to 15 kHz.

This allows the addition of more capacitance (100pF to ground) on the UI board, on the side of the added resistors that go to the RF board. This can further-reduce the one-second "tick" sound that some report on higher bands.

The update of firmware is automatically detected and new EEPROM variables automatically initialized:

Starting with this version, the "build" number ("211" in this version) is stored in the EEPROM and compared upon bootup. 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.

Available adjustment ranges and defaults of the DSP NOTCH configuration settings have been changed to more useful ranges and values:

A bit of experimentation has revealed that new default values for "**DSP Notch ConvRate**" and "**DSP Notch BufLen**" are better than the previous default values: These may be obtained by going to these settings and pressing the **F2** (DEFLT) button. The available ranges both of these settings has been changed to more reasonable values.

Version 0.0.209

December 20, 2014

Changes include:

- Fixed memory save of #28 (AM Disable) and #294 (20M, 5 Watt Adjust)
- "TX Disable" added
- Modified "temporary" tuning size to freeze tuning knob until change takes effect
- Frequency Tune Lock added
- Modified AGC to make audio "Smoother"
- Modified TX ALC to make audio "Smoother"
- Added I/Q gain balance for AM
- Added automatic switching of on-screen parameters between TX/RX
- DSP now works in 10 kHz mode

IMPORTANT:

> **When you install new firmware, FIRST, power-on the transceiver, and then power it off again using the POWER button.**
>
> **This is necessary to initialize the EEPROM locations used in the new variables in the new firmware. You need to do this only once after loading the new firmware the very first time.**
>
> **Remember also that for unknown reasons the DSP filtering will occasionally "crash." If you power up the mcHF and get either no audio or badly distorted audio when DSP is enabled, or if this occurs when you are listening, the only way currently known work-around is to disable DSP or power-cycle the transceiver. The cause/fix for this is currently being investigated.**

Details:

Fixed memory save of #28 (AM Disable) and #294 (20M, 5 Watt Adjust):

In a recent version the above two parameters got broken *(or maybe never worked)* – but they are fixed now. *(Thanks for spotting this, Tino!)*

"TX Disable" added:

A TX Disable feature has been added and this may be activated two in two different ways:

- **Press-and-Hold the TUNE button to enable/disable transmit**
- **Menu item 202 "Transmit Disable"**

Both methods of disabling the transmit do ***exactly*** the same thing. When the transmit is disabled, the **TUNE** is set to dark(ish) grey in color. *Note that menu item numbers may change in future firmware releases as additional menu items are added.*

Modified "temporary" tuning size to freeze tuning knob until change takes effect:

In version 0.0.208 a "temporary" step size change function was added in which pressing-and-holding the **STEP-** or **STEP+** button would *temporarily* change the step size, allowing one to more easily fine-tune a signal or make large frequency changes, respectively.

What has been added is the "freezing" of the tuning control at the instant that one presses the button and persists during the "hold" time, expiring when the "press-and-hold" detect time has elapsed and the temporary step size has taken effect.

In version 0.0.208 if you pressed-and-held the button and started turning the tuning knob immediately, the original step size would be in effect for a second or so before the "press-and-hold" step size took effect. For example, if you were pressing-and-holding the **STEP-** button to fine-tune a signal, this could cause one to mis-tune a signal if one were not very careful to wait for the press-and-hold time to expire and the step size to change.

Now, this will not happen since you will not be able to tune until the change has taken effect!

Frequency Tune Lock added:

The frequency tuning knob may be locked/unlocked by pressing-and-holding both the **STEP-** and **STEP+** buttons at the same time with the frequency lock being indicated by the frequency display turning grey – but note that the **RIT** function will still work. It should be noted that when pressing or releasing these two buttons that the step size may change if you do not press/release them at precisely the same time.

Modified AGC to make audio "Smoother":

The receiver's AGC function has been reworked and listeners – particularly those that use CW – should notice that the receiver will now sound much "smoother" (e.g. less "clicky"), especially with strong signals and if you use a fast AGC. Among the techniques applied are an "AGC Memory" and a slight audio delay added so that the action of the AGC occurs precisely in unison with the audio, (almost) completely preventing overshoot/undershoot that would otherwise occur with fast attack and decay.

Modified TX ALC to make audio "Smoother":

The transmitter ALC has been modified in the same way as the receiver AGC which also improves its "sound".

Added I/Q gain balance for AM:

A new parameter, configuration menu item #244, "**AM RX IQ Bal.**" has been added. This adjusts the receiver I/Q amplitude balance when in AM mode and is used to minimize the low-level "tweet" (e.g. tone) that may be heard when an AM signal is tuned in slightly off frequency to avoid the "zero Hertz" hole.

Note that menu item numbers may change in future firmware releases as additional menu items are added.

This frequency of this "tweet" is twice the offset from the carrier frequency, which is to say that if you tune 500 Hz from the AM carrier, you will hear a 1000 Hz tone. 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 the resulting 1000 Hz tone.

While this adjustment is unlikely to completely eliminate this "tweet", it can significantly reduce it. Note also that the efficacy of this reduction changes somewhat 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.

Added automatic switching of on-screen parameters between TX/RX:

The on-screen menu settings for "**AFG**" and "**RIT**" in receive will now automatically change to "**CMP**" and either "**MIC**" or "**LIN**" in voice modes or "**STG**" and "**WPM**" in CW mode, respective, when you go to transmit.

This was done to allow easier adjustment of parameters such as microphone gain and speech processing when using SSB, or adjusting sidetone gain and sending speed in CW while transmitting without having to pause and use buttons “**M1**” and/or “**M3**” to change their modes. Note that the previous mode of these indicators will be “remembered” between TX and RX so if, for example, you already had it set to adjust “**CMP**” in SSB mode during receiver, it would still be set thusly after you returned from transmit mode.

This function may be disabled using parameter #203, “**O/S Menu SW on TX**” which is an abbreviation for “On Screen Menu Switching on Transmit”.

Note that menu item numbers may change in future firmware releases as additional menu items are added.

DSP now works in 10 kHz mode:

Both DSP noise reduction and Automatic Notch Filter now work in the 10 kHz mode. Note that the noise blanker is disabled when set to 10 kHz mode. Because of the heavy processor loading, expect the user interface – particularly the Spectrum Scope – to slow down, particularly if both the DSP Noise Reduction (NR) and the Notch are enabled.

Additional comments:

- Keep in mind that the DSP and Noise blanker consume significant processor time and that turning these functions on can cause the reaction to the user interface to slow down. The most notable effects of this are sluggish response to the pressing of buttons and the slowed display updates of the Spectrum Scope. Note that the DSP NR (Noise Reduction) and the “Notch” are *separate* functions and each one, by itself, will consume processor power.

- The Noise Blanker is disabled in 10 kHz mode and in all AM bandwidths.

- In a change from 0.0.208 to 0.0.209, the 10 kHz bandwidth mode uses decimation-by-two in the receive audio processing to reduce processor loading and permit the availability of DSP. Because of this it has somewhat minimal low-pass filtering of the alias response on the audio output which means that there can be significant energy above 13-15 kHz under some conditions.

Version 0.0.208

December 14, 2014

Changes include:

- Changed main display "CMP" setting to adjust speech processor strength
- Button "F2" now just says "Meter" rather than changing with meter modes.
- Added "press and hold" capability to buttons to enable additional UI features.
- DSP Noise Reduction strength now adjustable from the main screen, selecting between NB and DSP.
Please read!
- Line-In/Mic selectable from main screen
- PREV/NEXT menu buttons may be used to jump to begin/end of menus
- Parameters may be saved to EEPROM at any time
- DSP On/Off/mode switch operation changed
- LCD Backlight brightness/dimming now available
- On-the-fly "temporary" tuning step size now possible

IMPORTANT:

> **When you install new firmware, FIRST, power-on the transceiver, and then power it off again using the POWER button.**
>
> **This is necessary to initialize the EEPROM locations used in the new variables in the new firmware. You need to do this only once after loading the new firmware the very first time.**
>
> **Remember also that occasionally, the DSP filtering will "crash." If you power up the mcHF and get either no audio or badly distorted audio when DSP is enabled, or if this occurs when you are listening, the only way currently known work-around is to disable DSP or power-cycle the transceiver.**

Details:

Changed main display "CMP" setting to adjust speech processor strength:

In this version, the "CMP" setting on the main display now interactively adjusts both the "**ALC Release Time"** and "**TX PRE ALC Gain**" parameters to provide a one-control adjustment of the "strength" of the TX speech processor.

A setting of 0 offers very little processing while a setting of 12 (maximum) is very "strong" speech processing. A value of 2 is suggested as being the default when little/no obvious speech processing is desired, but reasonable, automatic tracking of varying microphone levels is desired.

In the main menu the on-screen "**CMP**" value corresponds to the new parameter called "**TX Audio Compress**". When this is set to a value other than "SV" *(see below)* the "**ALC Release Time**" and "**TX PRE ALC gain**" parameters will be displayed in red and not be available for adjusting.

Custom Speech Processor settings:

Above 12 *(what would be a value of 13)* a setting of "**SV**" is displayed, which refers to "Selected Values". In this mode you can custom-select your own parameters for configuring the speech processor using the "**ALC Release Time**" and "**TX PRE ALC gain**" parameters. These parameters are *not* overwritten when a "pre-set" compression level is used.

Button "F2" now just says "Meter" rather than changing with meter modes:

This button used to change to “Audio”, “ALC” and “SWR” depending on the mode, but it now just says “Meter”. This was done because the indication was redundant, anyway!

Added “press and hold” capability to buttons to enable additional UI features:

The keypad interface code was significantly reworked, allowing greater flexibility including the ability to handle “press-and-hold” key conditions, allowing the potential for significant expansion of the user interface. The following features take advantage of this added capability.

DSP Noise Reduction strength now adjustable from the main screen, selecting between NB and DSP:

It may be noticed that to the right of “RFG” is the “DSP” setting, adjustable with **ENC2** (the middle of the three encoders) and selectable using button **M2** (below the encoder.)

If this button is pressed-and-held, one may toggle this display between “**DSP**” to adjust the “strength” of the DSP noise reduction (the same as the menu item “**DSP NR Strength**”) and “**NB**”, which adjust the “strength” of the noise blanker as before and is the same as the new menu item “**RX NB Setting**”.

Line-In/Mic selectable from main screen:

Pressing-and-holding button **M3**, below **ENC3** (the right-hand encoder) will toggle between **MIC Input “**MIC” and **Line Input “LIN**” modes**.** Note that this change will only actually occur during receive: If you are transmitting when you do this action you must briefly un-key for it to take effect. This is the same as menu item “**Mic/Line Select**”

PREV/NEXT menu buttons may be used to jump to begin/end of menus:

When in the **MENU** mode, **pressing-and-holding** the **PREV** or **NEXT** buttons will allow one to instantly jump to the beginning or ending of the menu.

If the configuration menu ***IS*** enabled, starting from the beginning of the main menu, using the press-and-hold function of the **NEXT** button, the sequence will be:

- End of main menu
- Beginning of the Configuration menu
- End of the Configuration menu
- Beginning of the main menu

This feature allows one to more-quickly access the first and last items of the ever-growing menu system and the items in-between!

Parameters may be saved to EEPROM at any time:

Pressing-and-holding the **F1** button at any time ***(except during transmitting!)*** will save parameters to EEPROM in the same way that a **POWER OFF** does.

If in **MENU** mode, the warning at the bottom of the screen indicating that settings should be saved will disappear – at least until something else is changed. If ***not*** in **MENU** mode the Spectrum Scope will be momentarily cleared and a message displayed.

NOTE: The EEPROM has a rated durability of “only” 10000 writes (minimum) so please avoid constant use of this function!

DSP On/Off/mode switch operation changed:

In this release pressing button **G2** will change the DSP mode. If in a voice mode, the sequence is:

NR → NOTCH → NR+NOT

While in CW mode, the sequence is:

NR → OFF (The notch filter is always disabled in CW mode!)

Now, a **press-and-hold** of button **G2** will toggle all DSP on and off, preserving the settings.

Press-and-hold of this button is the only way to turn off DSP noise reduction when in a voice mode!

LCD Backlight brightness/dimming now available:

A ***brief*** press of the **POWER** button will provide dimming of the LCD's backlight. There are currently four settings and pressing the button repeatedly will cycle through all of these settings:

- 100% (default)
- 50%
- 25%
- 12.5% (dimmest)

IMPORTANT NOTE:

The dimming of the backlight is done using PWM (Pulse-Width Modulation) methods and as such, it causes modulation of the +5 volt power supply.

> **Because of this, you should expect the dimming of the LCD to cause an audible tone in the receiver unless additional LCD power supply filtering is added! Use of the DSP "NOTCH" or "NR+NOT" modes may offer reduction of this tone but note that the DSP "NR" by itself may actually enhance this tone!**

The previous "board modification" files included the addition of a series resistor and a bypass capacitor on the LCD +5 volt line, but with PWM, this filtering is inadequate: The **current** version of the "board modification" document provides details on additional filtering that can greatly reduce the amount of tone that works its way into the receive audio.

On-the-fly "temporary" tuning step size now possible:

If the **STEP-** or **STEP+** button is press-and-held, you will notice that the step size – and the color of the numbers – will change will the button is being held. This allows the ***momentary*** change of step size to facilitate tuning in stations.

The use of **STEP-** will reduce the step size to allow fine-tuning of SSB/CW signals while **STEP+** will allow for much wider tuning steps to allow much faster tuning across the band. The sizes of these "alternate" steps depends on the initially-selected step size.

Version 0.0.207

December 7, 2014

Changes include:

- Fixed bug with TUNE in SSB mode
- RX Filtering improved
- Improved TX audio filtering
- "Max RX Gain" parameter added
- Dramatically improved operation/performance in AM (receive) mode.
- Added audio DSP noise reduction and automatic notch filter

IMPORTANT:

> **When you install new firmware, FIRST, power-on the transceiver, and then power it off again using the POWER button.**
>
> **This is necessary to initialize the EEPROM locations used in the new variables in the new firmware. You need to do this only once after loading the new firmware the very first time.**

Details:

Fixed bug with TUNE in SSB mode:

With the introduction of the ALC/Speech processor/compressor, a bug was introduced in the TUNE mode in which the power level was affected and there was a slight "motorboating". This has been fixed.

RX Filtering improved:

With the rewriting of the RX function the audio filters have been improved – most notably the CW filters which are dramatically "sharper" than before. This was made possible by the implementation of 4x decimation to reduce internal bandwidth for the filtering function.

Improved TX audio filtering:

The TX audio is now much better-filtered (almost "brick wall") with the audio bandwidth set to about 250-2800 Hz at the -6dB points, dropping off by 30dB below 225 Hz and above 3060 Hz. The TX audio filter has also been "tweaked" to compensate for the low-end rolloff of the Hilbert transformer so that it is now flat to within 1dB from 300 Hz to at least 2500 Hz.

The side-effect of this is that the opposite-sideband rejection on transmit is also very much improved.

"Max RX Gain" parameter added:

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. A setting of "0" corresponds with maximum gain and may result in an unstable receiver while a setting of "6" reduces the ultimate sensitivity of the receiver. This is subtly different from the "RFG" (RF Gain) setting internally.

Dramatically improved operation/performance in AM (receive) mode:

The performance in AM (receive) mode is now dramatically improved, but its operation requires some explanation due to the quirk common to many SDRs.

The "Zero-Hertz" hole problem:

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!

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

As noted in the current modification file, it is recommended that capacitors **C71** and **C73** 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.

How the demodulation is different from before:

In versions prior, the ***pre-demodulation*** bandwidth was fixed at 10 kHz, which meant that all signals within +/10kHz 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 +/-10kHz passband also be intercepted. The selectable audio filter was applied ***after*** the demodulation, but if there was an extraneous signal within the +/-10kHz passband, the damage was already done!

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

The AM bandwidth filtering operates as follows:

- **10 kHz:** Pre-detection bandwidth: +/-10kHz; Post-detection bandwidth: 10 kHz.
- **3.6 kHz:** Pre-detection bandwidth: +/-3.6 kHz; Post-detection bandwidth: 3.6 kHz.
- **2.3 kHz:** Pre-detection bandwidth: +/-2.0 kHz; Post-detection bandwidth: 2.3 kHz (300-2600Hz, adjustable).
- **1.8 kHz:** Pre-detection bandwidth: +/-2.0 kHz; 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 bandwidth appears to be a bit on the narrow side to accommodate the sidebands that extend out beyond the filter (e.g. greater than the +/-2kHz bandwidth).

The 1.8 kHz and 2.3 kHz filters are the same as those used for SSB and the center frequencies of these filters may be adjusted in the menu as desired.

Because it is always necessary to off-center tune an AM signal, 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.

Known issues with AM demodulation:

- 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 in the receive system: It **may** be possible to address this in the future, but for the time-being, off-tune the carrier until it disappears ***or*** you may try turning on the DSP notch filter.

Added audio DSP noise reduction and automatic notch filter:

In this version, a **PRELIMINARY** DSP noise reduction and automatic notch filter has been added. The addition of this feature has been made possible by the rewrite of the RX function to include decimation-by-four which dramatically decreases the required processor loading.

PLEASE READ THIS SECTION CAREFULLY!

But first, here 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, but it will work again when DSP is turned off. The cause of this is currently unknown and the only known "fix" (for now) is to turn the mcHF off and then back on again!

- If, when you turn on the mcHF there is no audio in one or more DSP mode, turn the mcHF off and back on again. Again, the cause of this is currently unknown and the only known "fix" is to turn the mcHF off and then back on again!

- If there is no audio when you turn the mcHF on, adjust the volume control: If "touching" the volume control fixes it, let me know. This is not DSP-related, actually, but I thought that I'd mention it, just in case.

The DSP mode is enabled/changed using button **G2**, the button that is the second from the left along the bottom edge and formerly used to change the keyer mode. There are four possible settings for DSP operation:

- **OFF**
- **NR** – Noise Reduction only
- **NOTCH** – Automatic Notch Filter only
- **NR+NOT** – Noise Reduction and Notch Filter

IMPORTANT OPERATIONAL NOTES related to DSP:

- **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" modes such as PSK31, RTTY, SSTV or any other digital modes.** ***DSP is NOT compatible with these modes!***

- **DSP is disabled in the 10 kHz bandwidth mode.** This is necessary due to the limitations of processor loading and the fact that decimation is disabled when operating at this bandwidth.

- **Enabling the noise blanker *and* DSP can cause the user interface of the mcHF to slow down significantly!** ***(You have been warned!)***

DSP Noise Reduction:

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

The "strength" of this filter may be adjusted using the menu item #10, "**DSP NR Strength"** - but be very careful with this as it easy to go overboard with this setting. If it is set too high, the artifacts caused by the noise reduction (e.g. "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.

Advanced configuration settings:

There are a number of "advanced" settings that you can play with that affect the DSP noise reduction, and these are located in the "Configuration Menu". These settings and their effects are somewhat technical in nature, so be thankful for the "**DEFLT**" button which will instantly restore them to their default settings if you mess things up!

- **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 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 voice too much and be ineffective. If we increase the delay, we can improve the performance but if we delay too much we end up with an "echo" type effect and a sluggish response. **This value must always be <u>larger</u> than "DSP NR FFT NumTaps", below. If this rule is violated, the number will turn RED and the DSP operation will become ineffective.**
- **DSP NR FFT NumTaps** – This is the number of taps in the FIR (filter) comprising the DSP noise reduction filter. A smaller number of taps implies a more agile filter, but also one that is less accurate while a larger number of taps is more precise and potentially slower to respond: A more "precise" filter may also reduce the actual performance in that the automatic calculation of the filter's parameters – which are, by their nature, imprecise, may "miss the mark". ***<u>A higher number will increase processor loading and slow the user-interface response.</u>*** **This value must always be lower than "DSP NR BufLen", above. If this rule is violated, the number will turn RED and the DSP operation will become ineffective.**
- **DSP NR Post-AGC** – This determines whether the DSP noise reduction will take place ***before*** the audio filtering and AGC or ***after*** the audio filtering and AGC. The net effect will be the same, but there will be important differences as perceived by the user:
  - **"NO": DSP Noise reduction takes place *before* filtering/AGC** – The operation of the DSP noise reduction **will** affect the S-meter reading. Because the noise reduction occurs prior to the AGC, the "quieting" caused by the noise reduction will be compensated by the AGC and the perceived "quieting" effect caused by the noise reduction will be reduced. **Note that this can give the *<u>impression</u>* that the noise reduction is less effective than it actually is!**
  - **"YES": DSP Noise reduction takes place *after* filtering/AGC** – The 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!**

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 Notch Filter:

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

The notch filter operates within the signal path ***prior*** to the AGC and the **DSP NR** operation, so a strong "tune up" signal will not cause the S-meter to deflect when the notch filter is active, but note that the 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!

Advanced configuration settings:

There are a number of “advanced” settings that can affect the DSP notch filter operation and these may be found in the “Configuration” menu, below the settings for the DSP noise reduction.

- **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.
- **DSP Notch 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 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 voice 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.

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!

Again:

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, but it will work again when DSP is turned off. The cause of this is currently unknown and the only known “fix” (for now) is to turn the mcHF off and then back on again!

- If, when you turn on the mcHF there is no audio in one or more DSP mode, turn the mcHF off and back on again. Again, the cause of this is currently unknown and the only known “fix” is to turn the mcHF off and then back on again!

- If there is no audio when you turn the mcHF on, adjust the volume control: If “touching” the volume control fixes it, let me know. This is not DSP-related, actually, but I thought that I'd mention it, just in case.

Version 0.0.202

November 22, 2014

Changes include:

- Bug in which the audio is muted when powered up until the audio (AFG) is adjusted is fixed.
- Problem of CW filters being disabled in LSB mode – even when set to “ON” in the menu – is fixed.
- Added menu option in “Adjustment” menu to swap the functions of the STEP- and STEP+ buttons.

IMPORTANT:

When you install new firmware, <u>FIRST</u>, power-on the transceiver, and then power it off again using the POWER button.

This is necessary to initialize the EEPROM locations used in the new variables in the new firmware. You need to do this only once after loading the new firmware the very first time.

Details:

Bug in which the audio is muted when powered up until the audio (AFG) is adjusted is fixed:

In order to fix another problem, I re-introduced a previous bug in which the audio was muted on power-up until the volume control was adjusted. This has been fixed.

Problem of CW filters being disabled in LSB mode – even when set to "ON" in the menu – is fixed:

It was noticed by Tino (after a couple of versions) that in LSB mode that the "Narrow" (CW) filters (e.g. 300 Hz, 500 Hz) could not be selected in LSB mode, even when the parameter "CW Filt in SSB Mode" setting was set to "ON". The problem was a set of missing parentheses!

Added menu option in "Adjustment" menu to swap the functions of the STEP- and STEP+ buttons:

With the addition of the (optional) marker line underneath the frequency display to indicate the step size, another menu item (#202 in this version), the **"Step Button Swap**" setting was added. When set to **ON** this will cause the functions of the **STEP –** and **STEP +** buttons to be swapped so that the left/right sense of the two buttons correlates with the movement of the marker line.

Version 0.0.201

November 20, 2014

Changes include:

- Noise blanker has been modified.
- Audio muting in RX->TX fixed.
- Optional marker below digit denoting step size.
- On-screen WPM/STG indicators now mode-context sensitive.
- Added ALC (Automatic Level Control) to the transmitter in voice modes.
- Added function to button F2 to switch between SWR, TX audio, and ALC metering – ***see below for a more thorough explanation.***

IMPORTANT:

> **When you install new firmware, first, power-on the transceiver, and then power it off again using the POWER button.**
>
> **This is necessary to initialize the EEPROM locations used in the new variables in the new firmware. You need to do this only once after loading the new firmware the very first time.**

Details:

Noise blanker has been modified:

The noise blanker function, which was experimental before, has been revised somewhat and should (in theory) be more effective. As before, the number will change color as a warning to the user that increased levels are likely to cause audio degradation.

Important note: The noise blanker takes a fair amount of processor horsepower and users will note that when activated (e.g. "NB" is set to a value ***greater than 0***) some operations of the transceiver (e.g. change of band, etc.) will be slightly sluggish, and the noise blanker is disabled when the menu is entered. In a near future revision, the receiver processing algorithm will be rewritten to dramatically reduce CPU load and this problem should go away. If this bothers you, just leave the noise blanker off!

Audio muting in RX->TX fixed:

It was observed that the audio input to the codec (e.g. U3) was being switched by the PTT line before the codec inputs were muted, causing significant disruptions when going from receive to transmit: This problem has been addressed.

Optional marker below digit denoting step size:

Menu item 201 (the second item in the "Adjustment" menu), when set to ON, enables the display of a bar under the digit denoting the currently-selected frequency step size.

On-screen WPM/STG indicators now mode-context sensitive*:*

When in CW mode the "**STG**" and "**WPM**" indicators display the "Sidetone Gain" and the Keyer Speed in Words-Per-Minute. In audio modes (e.g. USB/LSB) these indicators are irrelevant so they now change to "**CMP**" and "**MIC**" or "**LIN**" respectively -The latter setting is the Microphone or Line gain, depending on which is enabled in the main menu. *See below for a discussion of the "**CMP**" parameter.*

Added ALC (Automatic Level Control) to the transmitter in voice modes:

An Automatic Level Control has been added to the transmitter in voice modes.

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 mcHF was somewhat irrelevant when in a voice mode as it only added effective attenutation 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 has been changed in this version of code: **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: If you do not understand its significance, please bring it up as a topic for discussion on the Yahoo group!

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.

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 still does not work... yet...***
- The **AUD**io meter. This shows the audio level from -20dB to +12dB, with 0 dB being "nominal".
- The **ALC** meter. This shows the amount of ALC action, from 0 to 34 dB – *more on this below.*

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

- Speak normally if using the Microphone input, or set the nominal input level if you are using the LINE Input mode.
- Use button **F2** to select the **AUD**io meter.
- Use button **M1** (below **ENC1**, the left-hand encoder) to select the on screen **CMP** setting and use that encoder to adjust it to a setting of 1.
- Use button **M3** (below **ENC3**, the right-hand encoder) to select the on-screen **MIC** (or **LIN**) setting and use that encoder to adjust it, or you may go into the Menu mode and adjust the "**Mic Input Gain**" (or "Line Input Gain" as appropriate).
- While speaking normally, adjust the gain so that the audio meter peaks up to "0" (zero) 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:

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

SSB operation with minimal speech compression:

- Set the Microphone/Line gain as described in the previous section *(e.g. around "0" on the **AUD**io meter.)*
- In the menu system, set the parameter **ALC Release Time** to the default setting of **10**.
- While speaking normally, use the **CMP** setting (a.k.a. "TX Compress Level") 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 **AUD**io meter.)*
- In the menu system, set the parameter **ALC Release Time** to the default setting of **3 or lower**.
- While speaking normally, use the **CMP** setting (a.k.a. "TX Compress Level") for a peak reading on the **ALC** meter of 8-16 dB.
- Setting **ALC Release Time** to a lower value and the **TX Compress Level** 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 Compress Level:** This is a variable audio gain ***after*** audio filtering in the transmit bandpass, ***after*** the audio metering, above, but ***before*** the ALC circuit.
- **ALC Meter:** This indicates the amount of gain ***reduction*** in deciBels that the ALC is providing to the audio path. The ALC is in the audio path ***after*** transmit audio filtering so it will not respond to audio that is outside the frequency range that will be transmitted. The ALC can only ***reduce*** gain *(by up to 40 dB)* but it can <u>never</u> increase it and it will settle to unity under no-signal conditions.
- **ALC Release Time:** This sets the time, after audio has dropped below the current threshold, that the ALC will take to release and reduce attenuation. When set to the default setting of 10, the ALC will have only a modest effect on the transmitted audio, taking several seconds for the ALC to completely recover from a voice peak while setting it to the maximum value if 20, the effect is almost that of disabling the ALC entirely in terms of added compression in that the gain recovery rate is approximately 1dB/second. Low values (below 5) will "follow" audio very quickly and offer effectively very high compression rate.

Comments on SSB operation:

Once you have properly set up the microphone/line gain to suit your taste, you need not readjust it when you change the power level as the **PEP** output, ***when you have indication on the ALC meter*** should be the same power level as the carrier output power that you get when you "key down" in CW or TUNE mode.

Warnings:

- Do not set the Mic/Line gain such that the peak audio level on the **AUD**io meter regularly peaks much above 4 to 8. Avoid settings that "peg" the meter as this could indicating clipping.
- Frequent, very high indications on the **ALC** meter *(e.g. >12dB)* 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.
- 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!

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, 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 0.1 to 0.3 amps.
    - Unkey the radio.
    - Use the **POWER** 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!

Version 0.0.200

November 15, 2014

Changes include:

- Fixed bug that prevented the save of the "Off" mode of Spectrum Scope
- Added the selection of Fahrenheit readout of TXCO temperature-based
- Fixed "glitch" in CW mode that occurred at the beginning of the first "Dit" element of a string of CW elements.
- Fixed problem in which TX to RX turnaround time was not consistent at the end of the last Iambic CW element.
- Added Spectrum Scope "AGC" adjustment.
- Added Spectrum Scope "Rescale" adjustment.
- Changed TX-to-RX timing to be based from hardware-interrupt source.
- Added muting of audio and "freezing" of AGC during band change to reduce "pop" and S-meter

IMPORTANT:

> **When you install new firmware, first, power-on the transceiver, and then power it off again using the POWER button.**
>
> **This is necessary to initialize the EEPROM locations used in the new variables in the new firmware. You need to do this only once after loading the new firmware the very first time.**

Details:

Fixed bug that prevented the save of the "Off" mode of Spectrum Scope:

This has been fixed!

Added the selection of Fahrenheit readout of TXCO temperature-based:

For the benefit of us backward Americans who still use Fahrenheit, or if you are nostalgic, you can now make the TCXO display read in this temperature scale.

Fixed "glitch" in CW mode that occurred at the beginning of the first "Dit" element of a string of CW elements:

Previously, there had been a "glitch" of extremely high power at the beginning of the ***first*** dit, when in Iambic mode, when entering transmit mode. This appears to have been due to a problem with synchronization of variables within the various state machines.

Fixed problem in which TX to RX turnaround time was not consistent at the end of the last Iambic CW element:

There was a problem in which the control of the TX to RX turnaround was based on the initiation of the element (e.g. the closure of the contacts of the key/paddle) rather than the actual instant at which the element was completed. This has been fixed.

Added Spectrum Scope “AGC” adjustment:

This allows the adjustment of the Spectrum Scope's AGC, the rate at which it changes with response to changing signals. A lower value = faster response.

Added Spectrum Scope “Rescale” adjustment:

This allows the adjustment of the rescaling of the signals within the displayed span of the spectrum scope. A lower value = faster response.

Changed TX-to-RX timing to be based from hardware-interrupt source:

The source of timing for the TX to RX turnaround is now based on a hardware interrupt and should now be more stable than before, but it was unlikely that variability would have been noticed unless one had turned off the spectrum scope.

Added muting of audio and “freezing” of AGC during band change to reduce “pop” and S-meter:

There had been a loud series of clicks and “pops” when the band was changed, along with a deflection of the S-meter. Now, the audio is muted to spare the ears of those wearing headphones and the S-meter is “frozen” during a band changed to minimize this disruption.

Version 0.0.199

November 8, 2014

Changes include:

- PA Bias adjustment range expanded
- Separate PA bias capability added for CW mode
- Slightly different grey/blue color on the S/Power meter to enhance visibility
- Reduced duration of muting during frequency tuning
- Added menu selection to disable AM
- Added menu selection to disable "Wide" bandwidth modes while in CW mode
- Minimum power factor in 5 watt and FULL power modes reduced from 5 to 3
- Additional qualification for interrupts to deal with apparent hardware bug in the MCU

IMPORTANT:

When you install new firmware, first, power-on the transceiver, and then power it off again using the POWER button.

This is necessary to initialize the EEPROM locations used in the new variables You need to do this only once after loading the new firmware the very first time.

Details:

PA Bias range expanded:

The way that the PA bias is calculated internally has been changed and the range of valid values has been expanded. Previously, the voltage range was from 2.5 to 4.4 volts (menu setting of 0-70). Because the bias voltage range of the RD16HHF1 FETs used for Q5 and Q6 can vary over a quite a range as specified by the data sheets, this has been expanded to the range of 1.35 to 4.5 volts (menu setting of 0-115.)

NOTE:

To convert the "old" bias setting to the new one, add 37 to the old value.

Separate PA bias capability added for CW mode:

There is now a separate PA bias setting in the "Adjustment" menu labled "CW PA Bias." As the menu label implies, this setting will be applied to the PA bias **ONLY** in CW mode, **ONLY IF THIS VALUE IS GREATER THAN ZERO.** This was done as a request since CW operation does not need the same degree of linearity as SSB and operating current may be saved by reducing the PA Bias.

This is currently an **EXPERIMENTAL** feature and some caveats apply:

- **Make sure that you have read the notice, above, about the change of the way the PA bias is calculated, and about how to convert the "old" values to the new ones!**
- There is currently no way to key the transceiver in CW mode without outputting RF energy. If you want to measure the PA bias current, you will have to make such measurements using the other PA Bias setting in SSB mode, keying the transmitter with no audio, and then setting the CW PA Bias to the same value.
- 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 output.

Slightly different grey/blue color on the S/Power meter to enhance visibility:

The blue signal level bar is slightly paler and the grey “unlit” portion is somewhat darker for better contrast.

Reduced duration of muting during frequency tuning:

To prevent occasional, loud “clicks” during tuning, the audio is muted during. The duration of this muting has been reduced. These clicks are due to the nature of the way the Si570 (U8 on the RF board) synthesizes signals and is tuned and the need for it to be one of its internal divisors be readjusted, which causes a brief interruption of its RF output every few hundred kHz.

If you were to disconnect the antenna and tune up and down on 40 meters you will still hear this occasional “click”, particularly when narrower bandpass filters are used. Unfortunately, the nature of the universe dictates that with a narrower filter, the duration of this “click” will be extended so it will be noticed more, while tuning, with the narrower filters. It will be barely noticeable – if at all – when an antenna is connected.

This “click” can also cause a momentarily deflection in the AGC as well, so while tuning, the AGC is value is saved and then restored, minimizing the degree that this “click” might cause a brief desense of the radio when this tuning.

Added menu selection to disable AM:

In the menu there is now a selection that allows one to disable AM from the selection of modes. Note that if you already have AM selected, it won't de-select it, but it just won't “come around” again when you make the mode selection.

Added menu selection to disable “Wide” bandwidth modes while in CW mode:

The menu now includes a selection that, when in CW mode, prevents the selection of the 3.6 and 10 kHz filters. If you already have one of these filters selected, already being in SSB mode or setting this menu item to disable it while in CW mode, it will not de-select it, but it just won't “come around” again when you make the filter mode selection.

Note that the 1.8 and 2.3 kHz filters are still available in CW mode if you disable the “wide” filters.

Added menu selection to disable “Narrow” bandwidth modes while in SSB mode:

The menu now includes a selection that, when in SSB mode, prevents the selection of the 300 and 500 Hz filters. If you already have one of these filters selected, already being in CW mode or setting this menu item to disable it while in SSB mode, it will not de-select it, but it just won't “come around” again when you make the filter mode selection.

Minimum power factor in 5 watt and FULL power modes reduced from 5 to 3:

It was reported by some users that a minimum power factor of “5” was not low enough to set the output power as low as 5 watts on low frequency bands (e.g. 80 meters) so the minimum is now 3.

Important:

The risk of reducing this power factor is that this diminution of output voltage from the codec starts to push the lower edge of the D/A converter resolution and there is the risk of increasing quantization noise during transmit, resulting in distortion and signal degradation. A much preferred solution would be to add low-frequency emitter degeneration to driver transistors Q3/Q4 that would reduce their gain at the lower operating frequencies.

Additional qualification for interrupts to deal with apparent hardware bug in the MCU:

A problem that had been identified – and was believed to have been solved – appeared again in this version, and that is the spurious triggering of a hardware interrupt from a completely unrelated I/O pin. In this case, the pins used for the STEP+ and STEP- (e.g. frequency step size), which had been known to cause spurious interrupts, seemed to be corrupting the receive function – specifically the AGC. This would cause the receiver to mute momentarily – or just go deaf – when the frequency step size was changed. It is entirely possible that other I/O inputs from encoders or buttons were triggering this or other interrupts and causing “odd” things to happen occasionally as well!

While it is possible to assign certain I/O pins to various interrupts, it is, according to the documentation for this ***NOT*** possible to assign these particular pins! (e.g. the Step+/Step- buttons.) At the present time there appears to be no errata describing this behavior.

What has been done is a much more thorough “qualification” within all known interrupt service routines. What this means is that when an interrupt is entered, the pin that is assigned to that interrupt is examined to see if it is in the state that is to have caused that interrupt. If it is, the interrupt is service, but if it is not, the interrupt flags are reset and the interrupt is exited.

In examining the code I could find only three relevant interrupts: One for the DAH/PTT input, which is the one that seemed to be being triggered in this case, one for the DIT input, and another for the POWER button. All three of these were re-written to assure that the states of the relevant input pins were properly qualified and that their flags were reset – something that I'd noticed was possibly not the case!

Version 0.0.198

November 4, 2014

Changes include:

- Modification of volume control to reduce step size and make adjustment less "touchy". Adjustment range is now 0-30.
- Addition of PREV/NEXT buttons on menu to move +/- 6 menu items to make navigation easier
- The addition of a "2x magnify" mode to the spectrum scope to display just +/- 12 kHz.
- Minor tweaks in the way the automatic compensation of the sidetone level works.
- Adjustment of 5 watt power level for each band.
- Adjustment of "FULL" power level for each band.

Details:

Modification of volume control to reduce step size. Adjustment range is now 0-30:

The amount of loudness change per step of the volume control has been reduced to make it less "touchy" and abrupt in its response. Where volume steps below 10 were solely handled by the codec chip, this threshold has now been raised to 16, and steps above this point are also less abrupt.

NOTE:

- **<u>You will probably want to increase the MAX Volume setting in the Adjustment menu!</u>**

Addition of PREV/NEXT buttons on menu to move +/- 6 menu items:

When in MENU mode you will note the addition of **PREV** (e.g. "previous") and **NEXT** buttons. Because the menus are getting rather large, these buttons were added to allow the 6 menu items at one time (e.g. one screen) to help navigate the system more quickly.

You will note that when you get the first menu item (using PREV) or the last menu item (using NEXT) it will always stop at that first/last item. This is done to assure that one always sees the first or last menu screen. Note that the "last" item of the menu system will depend on whether or not the "Adjustment" menu is enabled.

Note on the use of the PREV/NEXT buttons:

- Because the **<u>last</u>** screen of the ***main*** menu has fewer than 6 items on it, it is possible (or even likely) that the PREV/NEXT buttons will ***skip*** over this screen, so you may have to use the knob (#2) to go forward or backward to select these items.
- You will note that the quickest way to get to the "Adjustment" menu the first time will be to use the PREV button after going into the menu, which will get you immediately to the last screen, turning on the the "Adjustment" menu, and then using knob #2 and/or the NEXT/PREV buttons to go from there.

The addition of a "2x magnify" mode to the spectrum scope to display +/- 12 kHz:

This option changes the span of the Spectrum Scope from +/- 24 kHz when OFF to +/- 12 kHz when ON with the bottom frequency scale reflecting this change. The spectrum scope's AGC operates only on signals within the range being displayed in either mode.

This mode does not increase the resolution of the spectrum scope, but rather doubles the width of the vertical lines.

Minor tweaks in the way the automatic compensation of the sidetone level works:

Improvements were made in the way the automatic volume adjust for the sideone is handled when in CW and TUNE mode when the power output is adjusted. There are still some slight "issues" when the power is adjusted toward the extremes of high and low power in that the amplitude does change slightly, but it should track better, requiring less "tweaking" of the "STG" (sidetone gain) adjustment when the power level is adjusted.

Adjustment of 5 watt power level for each band:

There is a power calibration factor for each band available to allow the setting at the 5 watt level.

Important notes on the 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.**
- Note that the 2 watt, 1 watt and 0.5 watt levels are based on proportional scaling of the 5 watt settings.
- You **<u>must</u>** be set to the band being adjusted!
- You **<u>must</u>** be set to the "5 watt" mode for the specific band being adjusted.
- 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.
- If ***both*** the band **and** 5 watt mode are not set properly, the band will be "oranged out" and you will not be able to adjust it – this being done to prevent accidental adjustment of the wrong parameter.
- **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 "mcHF board modifications" document for updates on this topic.***

Adjustment of "FULL" power level for each band:

There is a power calibration factor for each band available to allow the setting at the "FULL" power level.

Important notes on the "FULL" power adjustment settings – <u>PLEASE READ CAREFULLY</u>:

- **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.**
- The "FULL" power setting has NO effect on any other power setting.
- You **<u>must</u>** be set to the band being adjusted!
- You **<u>must</u>** be set to the "FULL" mode for the specific band being adjusted.
- While you may 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.
- If ***both*** the band **and** FULL power mode are not set properly, the band will be "oranged out" and you will not be able to adjust it – this being done to prevent accidental adjustment of the wrong parameter.
- Note that "officially" the mcHF 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 <u>not</u> 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, so please take care when adjusting the "FULL" power parameters!
- ***Please follow the discussions on the Yahoo Group and check the "mcHF board modifications" document for updates on the topic of improving the power amplifier of this radio!***

Version 0.0.197a

November 2, 2014

Changes include:

- Minor bug fixes in CW mode to help un-tangle the state machine when going between TX and RX
- A new option for the TCXO On/Off mode is STOP which prevents polling of the temperature sensor and the 1-second "TIC" sound. (Ver. 0.196)
- The CAT button has been removed from the main screen and moved to the Adjustment menu. (Ver. 0.196)
- The Keyer Mode display has been removed from the main screen. (It is in the menu.) (Ver. 0.196)
- The CAL mode has been removed from the main screen and its functions moved to the Adjustment menu and separate phase/gain adjustments are now available for LSB and USB modes.
- The "crash" or pop that often occurs when going from transmit to receive in SSB mode, and the upward deflection in the S-meter in AGC has been quashed.
- Some of the alternate 2.3 kHz filters' bandpass responses were incorrect - they are now fixed
- A sidetone is now present when in TUNE mode when in SSB mode to indicate that the transmitter is outputting power.
- There is a calibration for the RF forward power meter in the Adjustment menu.
- There is a "Transverter" mode that allows the display to be offsetted for the local oscillator of an attached transverter.

Details:

Minor bug fixes in CW mode to help un-tangle state machine when going between TX and RX:

It was noted that the TX->RX turnaround applied to release of the paddle itself and not the completion of the CW elements. This was fixed, always enforcing a minimum TX->RX delay which helps assure that the various state machines get sorted out to reduce the probability of strange things happening (bleeps, clicks, state machine briefly "hanging", etc.)

These efforts seem to be largely effective, but there are some occasional oddities that still reported to occur.

STOP mode in the TCXO On/Off selection:

This stops the polling of U10, the MCP9801 temperature sensor that is used to display the temperature of the oscillator on the screen and provide frequency-temperature correction. When set to STOP mode, the word "STOPPED" will appear on the display.

This may be useful for those who have not yet modified their Ui boards by adding series resistors on the SCL and SDA boards and have the 1 second "Tick" on higher bands.

If you have not already done so, it is still recommended that you modify your Ui board, however, as the having the TCXO turned on can improve frequency stability!

CAT button removed from main screen, moved to Adjustment menu:

To make space for future options, the CAT function has been moved to the Adjustment menu. The state of the CAT button is NOT saved in EEPROM.

NOTE: It has been observed that if the CAT mode is toggled On/Off that one or more of the following may occur:

- If you turn CAT mode ON, have the USB cable connected and have recently programmed the radio, **IT WILL CRASH THE RADIO's SOFTWARE!** To prevent this, disconnect the USB cable after programming, power down the radio, wait 10 seconds, and then reconnect and power up again.

- If the CAT mode is turned ON and even if it is turned off again, the saving of EEPROM variables MAY NOT WORK when you power down using the POWER button. The reason for this is unknown.

The Keyer Mode display has been removed from the main screen:

This was done because it is a rarely-changed item (e.g. IAM_A, IAM_B, STR_K) and room was made for future features: Those who use CW do not often change the way the keying works, anyway.

It is now exclusively in the main configuration menu.

The CAL mode has been removed from the main screen and its functions moved to the Adjustment menu with separate gain/phase adjustments for LSB and USB modes:

You will now find that the Frequency calibration, PA bias, as well as the RX and TX IQ gain and phase adjustments are located in the "Adjustment" menu. The LSB and USB adjustments for both RX and TX have also been separated.

IMPORTANT:

- You should readjust your TX and RX gain and phase adjustments with this new firmware!

- Settings are NOT automatically saved: You must save settings by powering off the transceiver using the POWER button.

NOTE: CW uses USB for both transmit and receive. Because of the way that it is generated, there is no phase adjustment for CW transmit.

You will notice that certain numbers are in WHITE and others are in ORANGE, and these change based on mode (USB/LSB) and whether the transceiver is in receiver or transmit mode. When they are ORANGE, they CANNOT be adjusted - this to prevent accidental adjustment for a menu item NOT related to a current mode.

The PA bias may not be adjusted unless in transmit mode.

Recommended procedure for adjusting RX IQ gain and phase adjustments:

- Set the mcHF to LSB mode
- Set the AGC to FAST mode so that the receiver recovers more quickly from the "clicks" from the phase adjustments.
- Tune in a strong, constant signal. This could be a shortwave broadcast station or a signal generator.
- Tune the mcHF dial frequency 1 kHz above the carrier frequency to obtain a strong 1kHz audio note.

- Now tune the mcHF 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.
- 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.
- If you hear the "leakage", adjust the LSB RX IQ Bal. to minimize it.
- Once minimized using the RX IQ Bal., adjust the RX IQ Phase to further minimize the "leakage". Note that adjusting the phase will cause "clicking" which may upset the AGC/S-meter briefly.
- Once the LSB leakage has been minimized, repeat the above procedure in USB mode.

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

If you use the mcHF primarily for CW, use rather low frequency CW note 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 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:

- **Loosely** couple your transmitter to a receiver and computer with a waterfall display. Do NOT connect the mcHF transmitter to your receiver, but connect it to a dummy load and place a from your computer-connected receiver near-is the mcHF so that it gets adequate signal.
- Switch to LSB mode on the mcHF.
- Set the mcHF to 1 Watt mode.Switch to **USB** mode on the computer-connected receiver. (Yes, USB.)
- Tune both the mcHF and the computer-connected receiver to the same frequency.
- Press TUNE mode. You will hear a 750 Hz tone and on the waterfall display, see a signal 750 Hz below the mcHF dial display. If you adjust the LSB TX IQ Bal. you should see the signal 750 Hz above the mcHF dial display frequency go up and down. Null this upper frequency as much as possible. Unless this is nulled as much as possible, nulling of the Phase adjustment will not be possible.
- 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 screen to clear after each adjustment.
- Once the best phase null is obtained, go back and forth between the gain and phase for the best null.
- Press TUNE again to exit TUNE mode.
- Switch the mcHF to USB mode.
- Keep the computer-connected receiver on USB mode, but this time null out the 750 Hz tone BELOW the mcHF dial frequency.

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.

The "crash" or pop that often occurs when going from transmit to receive in SSB mode, and the upward deflection in the S-meter in AGC has been quashed:

When going from transmit to receive while in SSB mode, there was often a loud "pop" that disrupted the Spectrum display and caused the S-meter and AGC to "twang" and go momentarily deaf. This problem has hopefully been addressed in two ways:

- The Audio buffers, which are shared between TX and RX, are now purged upon returning from TX to remove TX content from them.
- The last RX AGC value is saved and re-used when returning to receive after transmit.

Some of the alternate 2.3 kHz filters' bandpass responses were incorrect - now fixed:

It was noted that some of the "alternate" center frequency 2.3 kHz filters were incorrect in their responses. These have now been fixed.

A sidetone is now present when in TUNE mode when in SSB mode to indicate that the transmitter is outputting power:

Almost all radios that produce a test carrier when in TUNE mode produce a tone to let you know that it is in transmit mode - and this one does now, too. The "test tone" frequency in TUNE mode for SSB modes is **ALWAYS** 750 Hz. When in TUNE mode set to CW the tone will vary with the Sidetone Frequency setting.

There is a calibration for the RF forward power meter in the Adjustment menu:

This adjusts the calibration of the RF "PO" meter and may be adjsted from 75 to 150, representing a multiplication of the original value by 0.75 to 1.50. The default is 100 (a multiplication of 1.00, or your radio's original setting.)

Note that the actual resolution of the bar graph power meter is fairly low!

There is a "Transverter" mode that allows the display to be offsetted for the local oscillator of an attached transverter:

In the Adjustment Menu there are two options:

- XVTR Offset Enable OFF/ON xN - When ON, the Offset is ADDED to the frequency display. This option also includes a multiplier (N) for the local oscillator that can range from 1 to 10.
- XVTR Offset (Hz) - This is the frequency, in Hz, that is ADDED to the frequency display AFTER the multiplication. The valid range is from 0 to 150 MHz, in 1 Hz steps.

The main display frequency is calculated as follows:

Display frequency = (LO Frequency * Multiplier) + XVTR Offset

NOTES:

- The SUB (smaller) frequency display remains un-multiplied, but the offset is added to it.
- When the XVTR Offset is turned ON, the frequency display is YELLOW to indicate that a transverter frequency offset is being added to the display.
- For frequencies above 100 MHz, the "100 MHz" digit is dropped. For example, "144.100.000" MHz would be displayed simply as "44.100.000". ***Be careful about this when you use the LO multiplier!***
- When setting the XVTR offset use the STEP- and STEP+ buttons - the same one that you use to adjust the tuning step size - to set the step size. Unfortunately there is no easy way to make 1 MHz adjustments in frequency at this time.

Again:

DO NOT forget to power off with the power button to save any changes that you make in any menus!

Version 0.0.195

October 26, 2014

Changes include:

- Problem of CW mode being "broken" after exiting TUNE mode now fixed, I think.
- Added multiple center frequencies for 300Hz, 500Hz, 1.8 kHz and 2.3 kHz filters.
- Renamed "2.6 kHz" to "2.3 kHz" filter as that is actually it's bandwidth!
- Added options to disable all filters, except the 2.3 kHz filter, if desired.
- Renumbered all menu items and changed various colors in the menu.
- CW Sidetone and TX offset range is now 400-1000 Hz, adjustable in 10Hz steps.
- Microphone Boost ("Mic Boost") is now automatic.

Details about the filter shifting and enable/disable:

In the menu you will find that everything has been renumbered, and the first items are those that relate to the audio filters, specifically the ability to shift their frequencies - or disable them entirely.

In the case of the 300 Hz and 500 Hz CW filters you will note that you can move their center frequencies in 50 and 100 Hz steps, respectively, from 500-ish Hz to 900-ish Hz (you can see for yourself) or disable them.

Note that when you disable them, if you have that filter selected, it will not immediately be disabled, but if you press the filter select button, you will find that it is no longer in the rotation.

You can also shift the center frequencies of the 1.8 and 2.3 kHz filters to suit your tastes.
You may also disable the 1.8 kHz, 3.6 kHz and 10 kHz filters. Because they are so wide, there is no real reason to shift the 3.6 or 10 kHz filters.

The DEFAULT filter selections are those that were original prior to this one (e.g. 0.0.194).

IMPORTANT NOTE ABOUT CW FILTER AND SIDETONE FREQUENCY SELECTION:

- MAKE CERTAIN that when you change the CW filter and/or the CW sidetone frequency that you make sure that each matches the center frequency and/or bandwidth of the other! If you do not, you may find that the other station comes back to you outside the filter bandwidth! This is particularly true if you use the 300 Hz filter and set the CW Sidetone below 500 Hz or above 900 Hz. ***You have been warned!***

Details about the CW mode being "broken" after TUNE mode:

This had to do with some of the settings not having been properly reset after TUNE mode was exited. As it turned out, if you had changed modes (e.g. out of CW mode and back again) then it would work again - which was a big clue as well.

This version also disables the MODE button and FILTER change buttons (e.g. G1 and G4, respectively) because changing these modes while in TUNE mode would have been just asking for trouble!

Other issues with CW mode:

There are reported occasional issues with the CW mode doing "odd" things. If you experience this, please note the conditions in which this occurs. Note that increasing the TX->RX delay may reduce the likelihood as it is suspected that these problems are most likely occur when the transceiver suddenly goes from TX to RX and then abruptly back to TX again.

Automatic Microphone Boost:

The "Mic Boost" item has been removed and made automatic: If the "Mic Gain" exceeds 50, the Mic Boost function is activated automatically and the range is now 1-100.

There is a slight bug in the display in which the “ones” zero of 100 is not being erased, but this will be fixed in the next release.

Various color changes:

You will notice various other color changes, mostly the item names and the options themselves, now being more consistent - that is, "default" or normal items generally being white, with other colors being warnings or, in special cases, color selections.

Firmware version 0.0.194

October 25, 2014

- The "RFG" control has been completely changed. Formerly, it adjusted the behavior of the Codec A/D converter gain scaling, but it now works like the "RF Gain" control on almost every HF rig you have ever seen: If set to zero, the S-meter will be "pinned" and the radio will be deaf, but if set to maximum (50) the radio will have maximum sensitivity. This control is used to set the "maximum sensitivity" of the radio and is useful on a noisy band to prevent the noise from coming up between transmission. Note that the numbers will go yellow, orange and then red as they are decreased to "warn" you that you are doing something that will "deafen" your receiver!

IMPORTANT NOTE: If, when you load this firmware, that you find your receiver to be "deaf", check the RFG setting. If it is Zero, set it 50 and all will be well - and be sure to power off using the POWER button to save the settings! ***(Don't say that I didn't warn you about this!)***

- STG Setting modified: The "STG" setting has now been modified such that the amplitude of the sidetone is now (nearly) constant, regardless of the TX power level.

- The problem with odd bleeps and blurps when using CW mode is fixed... I hope... See below about "CW TX->RX Delay". This should also reduce the problem with a "chirp" when going between RX and TX on SSB.

- The TX power coefficient for "FULL" power was reduced from 1.00 to 0.75 as it was observed that the original value was causing unconditional clipping of the TX QSD mixer, resulting in some dirty, across-the-band splattering. The peak-to-peak voltage being applied to U17's analog inputs was in excess of 7 volts, well above the "safe" limits of the device!

- The transmit filter has been replaced and it now better filters at communications bandwidth (e.g. below 3 kHz) as it should.

- A slight change was made in the way the TX IQ Gain balance works. Due to the intrinsic asymmetry in 0-90 degree Phase-Added Hilbert transformers, just one setting for one sideband does not work perfectly for the sideband the one for which the nulling was adjusted, so a work-around was added to provide an approximation for that opposite sideband. At some point separate TX IQ gain and phase adjustments for USB and LSB may be added... It is recommended that the TX IQ Gain be re-adjusted for best null with a compromise between USB and LSB.

- TCXO Button removed: The button that used to have "TCXO" on it is now blank. I haven't decided what to put there...

COMMENT on PA Bias Adjustment:

There seems to be a bit of confusion on the way that PA bias is adjusted. For now, this seems to be the best way:

First, it is strongly recommended that a resistor (1k-10k) be placed in parallel with C96 to assure that the PA Bias will adjust properly. If this is not done the bias can sometimes "run away" or simply not seem to have any effect as it is adjusted.

Second, it is strongly recommended that a tantalum capacitor of 4.7-22uF (exact value not critical) of at least 16 volts be placed across C106 to suppress spurious LF/MF oscillations.

(Both modifications are noted in the "Modifications" file in the KA7OEI filter)

When adjusting the bias, the following procedure is recommended:

- Set to 40 meters
- Set to CW mode
- Connect a watt meter
- Connect a dummy load

- Set to 1 watt mode
- Go to CAL mode
- Go to TUNE mode and adjust PA Bias for 1 watt of RF
- Exit CAL mode to save bias settings
- Exit Tune mode.

You should get about 1 watt (+/- 20%) on the other bands. If it's lower on some bands you may need to check the way the toroids for those bands were wound.

* * *

The first thing that you will note is that button F1, which formerly said "MIC" or "LINE" (or "MB" in "CAL" mode now says "MENU" when the radio starts up. When in MENU mode the CAL button is always greyed out - and the menu button will be greyed out when in CAL mode. Other than that, the radio should function more or less normally when in MENU mode.

Pressing this button will (surprise!) enter the main configuration menu. In this mode:

- Encoder 1 (far left) is ALWAYS the volume control.
- Encoder 2 (center) selects the item to be changed. A green " < " on the far right edge of the screen indicates the item being selected.
- Encoder 3 (far right) changes the item selected. CHANGES ARE IMMEDIATE!

When in MENU mode, the indicator above button F1 changes to "EXIT" and that above button F2 changes to "DEFLT" which means "DEFAULT" - which is to say that pressing that button will reset the selected item to its DEFAULT value.

If you change ANY item, TWO things will happen:

- The bottom of the menu will display, in orange, "Save settings using POWER OFF!" which means that NONE of the changes that you will make will be saved UNLESS you turn off the radio using the POWER button! If you simply remove the power, all changes that you make will be lost.

- When you exit the menu, you will notice that the "MENU" button is will now be orange with an asterisk next to it, reminding you that you may have changed something and that if you want to save it, you will need to power down the radio using the POWER OFF button!

(Note that it detects if you changed something, but it can't detect if you changed it back to what it was and then "un-indicate" that you didn't actually made a change.)

Note that, even while in the MENU mode you CAN power off the radio at any time and save settings.

* * *

Explanation of menu items:

The numbering of menu items are by general grouping which should be fairly self-explanatory. No doubt, things will be added as they occur to me and, possibly by request!

10 - **AGC Mode**: Slow, Medium, Fast, Custom and OFF (Default = Medium, which was the "fixed" setting in previous firmware versions.)

Slow, Medium and Fast are fairly self-explanatory, found on most other radios.

WARNING: If you select "OFF" you should turn the volume DOWN as you may get your ears blasted until you set the RF gain control (menu item #11, also adjustable from the "main" panel without going into the menu as noted above) adjusted downwards! **YOU HAVE BEEN WARNED!** *(This is one reason why the volume control is enabled when in MENU mode)*

11 - **RF Gain:** This is **exactly the same** as the now-changed "RFG" control on the main panel and it works the same way as RF gain controls on most radios. 0 = minimum sensitivity (Full-scale S-meter) and 50 = maximum sensitivity.

This may be used with the "AGC Off" mode if you wish to manually adjust the gain for some reason. It is most useful for listening to a noisy band to keep the background noise from coming up between transmissions.

In this menu the numbers do not change color as you decrease the setting and cause the receiver to go "deaf" - mainly, because I forgot in this version!

12 - **Custom AGC**: When the AGC mode is set to "Custom", this will set the "Hang" time of the AGC from very fast (0) to very slow (30). The higher values tend to make the audio sound quieter due to integration effects. Note that either extreme is likely to be unusable, but the choice is yours! The default value of 12 is equal to "Medium" AGC.

13 - **RX Codec Gain:** This is what had formerly been the "RFG" setting on the main screen. When set to Auto (recommended!) the A/D converter gain of the codec is automatically adjusted based on the signals that are present, decreasing the gain if excessively strong signals are detected. When this happens, you may occasionally see the bottom half of the S-meter, which is normally white, flash red, when an overload (or "near overload" condition is detected.

A setting of "8" is "Maximum" gain which makes the receiver VERY susceptible to overload on strong signals while a setting of "0" makes it comparatively deaf on a quiet band. If a setting OTHER than "AUTO" is selected the display is red with brackets to warn you that you probably should not be doing this!

* * *

20 - **Mic/Line select:** This selects the Microphone or Line input for voice/data transmissions

21 - **Mic Gain:** This adjusts the Microphone gain. The default (from previous code) is 5. **BE VERY CAREFUL** as there is no easy way to tell if you are overdriving the transmitter (at least, yet...) The number displayed is that which is used to multiply the signal from the microphone, which means that a setting of 1 is 1/5 of the voltage (-14dB audio reduction) and a setting of 50 is a 20 dB increase ***as compared to the "original" setting of 5***. **It is recommended that you do NOT set this above 10 or so if you enable Mic Boost**.

22 - **Microphone Boost:** When enabled, an extra 20dB of microphone amplification is enabled. **It is recommended that you do NOT set the Mic Gain above 10 if you enable Mic Boost!** ***The "Mic Boost" function may automatically be enabled and this setting removed in a later version of firmware!***

23 - **Line Gain:** This adjusts the gain when the LINE input is used. The default gain (from previous code) is 10. **BE VERY CAREFUL** as there is no easy way to tell if you are overdriving the transmitter (at least, yet...)

* * *

30 - **Keyer Mode:** This sets the Keyer Mode between Iambic A and B and Straight Key, the default being Iambic-B. This is the same setting as is on the main screen: It is expected that the setting on the main screen will be removed when that spaced is needed for additional features.

31 - **CW Keyer Speed:** This sets the keyer speed, the default being 20 WPM. This is the same as the "WPM" setting on the main screen. The default is 20 WPM.

32 - **CW Sidetone Volume:** This is the same as the "STG" setting on the main screen. The code has been modified so that the volume of the sidetone is more nearly constant with variations of the TX power level. The default is 5.

33 - **CW Sidetone/Offset Frequency:** This is the frequency of CW sidetone as well as the TX/RX offset - which is to say that if you were to tune a CW station's note to match that if the sidetone, your transmit frequency would be exactly the same as the other station's. The default is 750 Hz.

Currently, the range is adjustable from 600 to 900 Hz in 10 Hz steps, but **BE WARNED: The extremes of this range can put you outside the passband of the 300 Hz CW filter!**

34 - **CW Paddle Reverse:** When on, the "dit" and "dah" are reversed, the straight key is unaffected. Default = Off.

35 - **CW TX->RX Delay:** This sets the delay between the last CW element and the return back to RX mode, with the default setting of 8 which is approximately 1/2 second.

If there are continued problems with odd "squeaks" or "bloops" when going between TX and RX mode when in CW mode, try increasing this value slightly and then let me know.

* * *

60 - **TCXO On/Off:** This turns the TCXO frequency compensation on/off, the default being "On" if the temperature sensor is present.

* * *

70 - **Spectrum Scope Speed:** This adjusts the display rate of the spectrum scope, or disables it entirely. (Additional work needs to be done on this to make it "smoother".)

71 - **Spectrum Scope Filter:** This adjusts the "strength" of the filter used on the spectrum scope. The default is "4". A setting of "0" is similar to what was present on firmware version 0.0.181.

Those who are using LCD with and SPI interface should probably set this to 1 or 2.

72 - **Spectrum Trace Color:** You can set the trace of the spectrum scope to any of the available choices!

73 - **Spectrum Grid Color:** You can set the background grid of the spectrum scope to any of the available choices, or if you set it to "black", turn it off.

74 - **Spectrum Scale Color:** You can set the frequency scale along the bottom of the spectrum scope to any of the available choices, or you can set it to "black" to turn it off.

* * *

Calibration Menu:

At the very end there is another setting: Menu item "00" that defaults to "Off". If this is turned on another menu system is enabled.

Currently, there is only one item in this menu:

201 - **Maximum Volume:** This sets the maximum volume that may be set, which may be useful to those that use headphones. The default of 13, it is adjustable from 8 to 14. The color of the number changes to warn you that you have "decreased" the setting to increasingly lower levels!

* * *

To be added to the main and calibration menus:

- Per-band settings for the TX power coefficients.
-

- Other things...

Changes with version 0.0.191

19 October, 2014

Changes in brief:

A problem in which transmitted audio was being heard in the speaker has been fixed.

No other changes in this version.

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Changes with version 0.0.190

12 October, 2014

Changes in brief:

- TCXO - Temperature-based frequency control reworked.
- Improved reliability of saved and recalled settings in EEPROM.
- Band-Up/Down buttons now "wrap around"
- The sub-display (used for determining RX frequency when RIT is used) should no longer show an occasional spurious "10's" MHz frequency digit.
- The CALibrate display would occasionally be corrupted when changing between TX and RX.
- The handling of the input attenuator (in the A/D converter) is improved.
- RIT setting is no longer saved in EEPROM.
- The RF Bandpass/Lowpass filters are now automatically changed with tuned frequency.
- Displayed band reflects actual tuning frequency.
- Bug in TUNE mode with erratic power level in USB/LSB mode fixed
- Sidetone more consistent in TUNE/CW mode
- An *experimental*, impulse-type noise blanker has been added.

* * *

In detail:

TXCO Frequency control improved:

Using an insulated thermal container (e.g. small cooler) I heated and cooled the transceiver while noting the frequency of the transceiver at 14.000 MHz on a GPS-locked monitor, making note of the frequency differences. It was noted that the resulting "curve" appeared to be typical of that of an AT-cut quartz crystal with a "turnover" at around 57C, according to the temperature sensor.

As of the last version (0.0.189) the general frequency calibration had been reworked such that the "reference" frequency was 14.000 MHz, and the frequency-delta information was applied here as well, using a temperature of 42-43C as a "nominal" temperature since this is approximately that at which a thermally-bonded Si570 and temperature sensor will operate in a 25C room when the transceiver is NOT in a case.

The algorithm has been rewritten to take advantage of the full, fractional-degree resolution of the temperature sensor and interpolate the temperature-frequency offset table to absolutely minimize the "jumps" as compared to the original code.

Notes:

- It is necessary that the Si570 and U10 be thermally bonded. This is done by placing a small piece of aluminium or copper across the top of the two chips and epoxying it to BOTH chips so that they are at as close to the same temperature as possible at all times.
- Not all Si570 chips are created equal. In my transceiver the temperature compensation holds the transceiver to 1-2 Hz of the correct frequency on all bands over a wide temperature range. If it turns out that there are reports that others' Si570 chips are radically different it may be necessary to add a utility where one can manually enter calibration points into EEPROM memory - a future feature, obviously!

* * *

Improved reliability of saved and recalled settings in EEPROM:

Previously, some settings were not reliably saved/ recalled: This should be much improved now (e.g. band/mode/configuration.)

Note that the settings are saved when one properly uses the **POWER** button to shut off the radio. If you simply remove the power, most changes other than those made in the CALibrate mode *(e.g. frequency, mode, filter, power, etc.)* will be lost!

*** * * ***

Up/Down band buttons now "wrap" around:

On 10 meters, pressing "Band-Up" will now go to 80 meters, while on 80 meters pressing "Band-Down" will go to 10 meters.

* * *

The sub-display (used for determining RX frequency when RIT is used) should no longer show an occasional spurious "10's" MHz frequency digit:

The small frequency display, which is used for showing the actual receive frequency when the RIT is being used, would often show a spurious "10's" of MHz digit, particularly when the transceiver was powered-up on a band below 10 MHz.

This has been fixed, I hope!

* * *

The CALibrate display would occasionally be corrupted when changing between TX and RX:

Sometimes the numbers, particularly those related to the Frequency Calibration, would get corrupted when going between receive and transmit, when in CALibrate mode.

(Also, it now says "FreqCal" instead of "CalFreq".)

* * *

The handling of the input attenuator (in the A/D c onverter) is improved:

If you recall, it has been recommended with the newer firmware that one REMOVES transistor Q2 as far better dynamic range adjustment can be had by adjusting U1, the main Codec on the UI board. This change reflects a slight change in the way that dynamic is handled under large-signal conditions.

As of version 0.0.183, when there are very strong signals the lower half of the S-meter *(the portion from S0-S9)* will flash red and if the signal is not TOO strong, the gain of U1 will be adjusted internally when RFG is set to "AUT" mode: If it continually flashes red, this would be due to a very nearby transmitter, or one that comes and goes intermittently.

Remember: One may always manually adjust it, with 8 being the highest sensitivity, most susceptible to overload and 0 being least sensitive, but most resistant to overload.

This change is largely invisible to the user and is designed to make it bet ter "anticipate" a stronger signal when in "Auto" mode.

* * *

RIT setting is no longer saved in EEPROM:

I can think of no good reason why one would want to save the RIT setting in EEPROM - can you? (If so, let me know?)

* * *

The RF Bandpass/Lowpass filters are now automatically changed with tuned frequency:

The RF filtering (both the bandpass and lowpass) are now automatically changed as one tunes the dial frequency. The thresholds are as follows:

- Below 4.25 MHz: 80 meters
- At or above 4.25 MHz, but below 8 MHz: 40/60 meters
- At or above 8 MHz, but below 16 MHz: 20/30 meters
- At or above 16 MHz: 10/12/15/17 Meters

You will hear the relay(s) click as you pass through these frequency boundaries.

* * *

Displayed band reflects actual tuning frequency:

The small indicator to the right of the main tuning frequency display now indicates the band. When tuning WITHIN an amateur band, it will indicate that band (e.g. 80m, 17m, etc.) but if outside that band it will indicate "Gen" for "General Coverage." The band limits are based on the largest found in the world.

It should be noted that for this and for ALL previous versions of code that if you power down on a frequency OUTSIDE an amateur band, it will revert to the lowest frequency of that band on power-up as it will have detected an "out-of-band" frequency when checking the limits. This is inherited from the original code as it had to guarantee that it started up with a valid, amateur frequency on power-up. At some point, this "gene ral coverage" frequency will become an extra "floating band" (in addition to the existing bands) in much the same way that Yaesu and Kenwood handle such things.

In other words, this transceiver has always done this, but what, exactly, it is doing is more apparent, now.

* * *

Bug in TUNE mode with erratic power level in USB/LSB mode fixed:

I noticed, in testing, that the power level would vary when in TUNE mode when in USB/LSB with the "MIC" and "LINE" setting - an oversight when I added the facility to generate the built-in test signal to allow TX IQ Phase adjustment.

(It worked properly in CW mode, before.)

Important note:

- Make sure that you have placed a resistor of 1k-10k across C96, near U18 in the "PA Bias" section of the RF board as noted in the notes of 0.0.189 and the board modification notes.

* * *

Sidetone more consistent in TUNE/CW mode:

The sidetone in TUNE/CW mode would sometimes be muted.

There may still be a problem with inconsistent volume/sidetone gain level to be worked out.

* * *

An *experimental*, impulse-type noise blanker has been added:

You will noted that the "ATT" menu item has been changed to "NB". *Again, make sure that you have removed the JFET transistor Q2 from the RF board as it has been obsoleted by previous versions of firmware!*

This is a very simple impulse-type noise blanker, with "0" being "OFF" and "15" being maximum. You will also notice that, while adjusting, the numbers will go from yellow to red, indicative of the potential to cause signal distortion - some of which can be seen, in extreme cases, on the spectrum display. Needless to say, some settings in RED probably won't be all that useful!

Typical of this type of noise blanker, it will be affected by a strong, adjacent-frequency signal - something easily spotted on the Spectrum Scope. If the blanker is operating too aggressively you will see the noise floor on the Spectrum Scope rise up and/or signals get wider: If the effect on the received audio is objectionable, simply reduce the NB setting.

This is *NOT* a DSP noise blanker, strictly speaking, but rather a software implementation of a noise blanker that one might see on a receiver from the 80's or 90's. In testing around my shack, it does seem to work, but I don't actually have much in the way of annoying im pulse noise (fortunately for me!) to try it out on.

This noise blanker may or may not be useful to anyone and it may not even work - particularly with "version 1" of this blanker! Please let me know your observations - and whether or not it works at all for you! (This first version may be utterly useless!)

* * *

Known issues:

- Reflectometer/SWR meter does not work, yet. This is still in the works.
- There *may* be an issue with various "strangeness" after using the TUNE/CAL mode, particularly in the USB/LSB mode. If you notice this, first save your values by exiting CAL mode and then letting me know what you did/saw: A power off-on will fix it if “strange” things occur.
- If you have been updating the radio's firmware, pressing the CAT button will crash the radio. This is "fixed" by disconne cting the USB cable for 10 seconds after programming.
- I'm sure that there are "new" bugs that I've introduced!

Changes with version 0.0.189

5 October, 2014

Changes in brief:

- Frequency, Mode and Filter now saved on a per-band basis.
- Start-of-band frequencies and sizes have been adjusted to align with Regions 1 and 2.
- The TUNE mode now works for USB/LSB without needing external audio tone generator for TX IQ Phase adjust
- Keyer mode is now saved.
- Frequency can now be calibrated.
- Tuning range extended down to 1.8 MHz and up to 32 MHz
- Tuning scope screen update function more efficient: This should slightly help those with SPI-mode displays (e.g. HY28A and/or Version 0.1 boards)
- Tuning scope smoothing filter will be less aggressive when in SPI mode: This should slightly help those with SPI-mode displays.
- PA Bias is now properly loaded on power-up and calculated differently, with wider adjustment range.
- Delay on power-down (with countdown) to assure more reliable write to EEPROM when saving settings.
- Sub-display is now properly updated when RIT is set to zero.

*** * ***

Important note about saved settings:

BEFORE installing this firmware write down your TX and RX IQ gain and phase adjustments, your STG (Sidetone Gain), WPM, RFG and other settings.

To work around a "bug" in the virtual EEPROM, the addresses had to be shifted so previous settings will be disturbed!

YOU HAVE BEEN WARNED!

*** * ***

Changes in detail:

Frequency, Mode and Filter now saved on a per-band basis:

For each of the nine bands the frequency, mode and filter setting is saved on each band in nonvolatile memory.

Please note that if the radio was tuned to a frequency outside an amateur band, it will be stored as the bottom frequency of the closest amateur band. In the future, there may be a "floating" general coverage band as is found in recent vintage Yaesu, Icom and other brand radios.

NOTE: Settings will not be saved if you do not use the POWER button to shut off the radio!

Comment: Default memory loaded from EEPROM (e.g. no data stored there before) will be 25 Hz from the lower band edge - this is a debug tool that I'd put there to indicate when memory was being loaded from default that I'll remove on the next release. It only shows up if there was not a frequency stored previously.

*** * ***

Start-of-band frequencies and sizes have been adjusted to align with Regions 1 and 2:

All amateur bands now start on frequencies in common with Region 1 and Region 2, and the sizes of the bands are those found in Region 2, which are the largest. It is the "start + size" that sets the upper limit of the valid range used when storing memories to determine if it is within a valid amateur band.

The starting frequency and "size" of 60 meters has been set so that it should cover the frequency range over which such amateur allocations are typically found.

*** * ***

The TUNE mode now works for USB/LSB without needing external audio tone generator for TX IQ Phase adjustment:

In version 0.0.187 it was required that an external audio generator was required for TX IQ phase adjustment. Now, the transceiver will generate a tone internally, allowing TX IQ Phase adjustment.

TX IQ Phase adjustment can now be adjusted ONLY in TUNE mode when set to USB or LSB mode! The setting will be grayed out in ANY OTHER mode!

Remember: First, null out the unwanted signal using the TX IQ Gain, then use the TX IQ Phase - and then do it again a couple more times!

*** * ***

Keyer mode is now saved:

It wasn't before, but it is now!

*** * ***

Frequency can now be calibrated:

When in Receive mode you will now see a "CalFreq" setting, adjustable by the far-left encoder that adjusts the calibration of the transceiver frequency. At this time, the range is +/-2000. At 14.000 MHz each step of this calibration represents 1 Hz and is proportional to the *current* operating frequency.

Note: Do Not enable the TCXO at this time!

Procedure, to be done at "nominal" room temperature:

- Tune in a known frequency reference (e.g. time station such as WWV at 10 MHz).
- Set the mode to USB.
- Now tune the frequency reference exactly 1 kHz LOW.
- Using a program such as Spectran, Argo, Spectrum lab to monitor speaker audio or using a frequency counter connected to the speaker, adjust the calibration to achieve an audio tone of exactly 1 kHz
- Press CAL to make the calibration menu disappear. This will also save the setting to memory.

The TCXO needs to be revisited - a project for a later release of the code!

* * *

Tuning range extended down to 1.8 MHz and up to 32 MHz:

Not much to say about this other than at the low end of the frequency range, the synthesizer ***MAY NOT LOCK RELIABLY*** and that receiver sensitivity specifications are not guaranteed.

Even if the transmitter is found to function below 75 meters (3.5 MHz) or above 10 meters (29.7 MHz) ***do not count on the spectral purity of the transmitter to be adequate for legal operation. Additionally, transmitting outside the amateur bands may cause damage to the final transistors.***

<u>YOU HAVE BEEN WARNED!</u>

* * *

Tuning scope screen update function more efficient:

and

Tuning scope smoothing filter will be less aggressive when in SPI mode:

With version 0.3 boards and HY28B displays a parallel interface is used, but with HY28A displays or older boards, an SPI interface may be used - which is slower. Unfortunately, this means that the spectrum scope - which is very display-intensive - can be significantly slowed-down.

In this version I tweaked the function that updates the Spectrum Scope display so that this should speed up the display significantly: The SPI interface will still be slower than the parallel, but it should (hopefully) be a bit faster.

In the Spectrum Scope display there is also a "low pass" filter that smooths the readings. Now, when in SPI mode this filter is relaxed a bit so the display will (hopefully) respond a bit more quickly as well.

As the display becomes more "white" it (probably) slows down in SPI mode (someone could probably verify this?) One of the future enhancements will be to modify the Spectrum Scope display so that its display will be less prone to "creep up" and turn white during periods of no signal or weak signals - something that should help those with SPI interface displays.

* * *

PA Bias is now properly loaded on power-up and calculated differently, with wider adjustment range:

It was noticed that the PA bias wasn't be loaded from EEPROM on power-up. I also noticed that the voltage range wasn't quite wide enough to go from "cut off" accommodate the entire range required.

IMPORTANT NOTE:

It is STRONGLY recommended that a resistor (1k-10k) be placed in parallel with C96 on the RF board, near U18.

What I noticed was that there is no DC loading on the output of U18 and it would tend to "float" up and not regulate properly at the intended bias voltage. What this means is that the PA transistors may not be being biased properly or, worse, they may be biased on somewhat while in receive mode!

* * *

Delay on power-down (with countdown) to assure more reliable write to EEPROM when saving settings:

It was noted that the EEPROM write function was not being completed fully when being powered down, so an additional delay was implemented. Additionally, a power-down message and countdown was added as well.

*** * ***

Sub-display is now properly updated when RIT is set to zero:

The problem in which the sub-display - which is supposed to show the RECEIVE frequency when using RIT - is sometimes displaying garbage - is now fixed. This only happened when the RIT was set to zero.

* * *

Known issues:

- Reflectometer/SWR meter. This is going to take a bit of work.

- Varying sidetone levels in TUNE/CW mode. The sidetone varies with power level - possibly randomly - I will need to look into this.

- After exiting TUNE mode, sometimes CW mode (maybe SSB) may not work correctly. This is an intermittent problem, but "fixed" by turning the radio off (with the power button to save values!) and then back on again. I'll try to figure out what is going on here, but the problem is easily "fixed" with the power-cycle...

- The TXCO is not properly tracking temperature, and the "steps" are too large, as noted above.

- If you have been updating the radio's firmware, pressing the CAT button will crash the radio. This is "fixed" by disconnecting the USB cable for 10 seconds after programming.

- I am sure that there are "new" bugs that I've introduced!

Changes with version 0.0.187

- The bug in which there was a disruption when the STEP+ and/or STEP- button was pressed is now fixed. There is no need to remove the capacitor(s). (Please put them back to protect against RFI/static if you removed them!)
- A minor bug in the volume control in which two of the volume settings were the same is fixed.
- Problems with the audio un-muting in CW mode after TX have been fixed (mostly.)
- RX Phase adjustment has been added.
- TX Phase adjustment has been added - but it has effect ONLY in SSB mode!
- The TX/RX calibration settings now actively change with TX/RX mode, and the TX Phase adjustment is only possible in USB or LSB mode where it has an effect!
- The CW TX and RX frequencies are now aligned with each other, and they fall within the center of the 300Hz and 500 Hz wide CW filters. (Sidetone/offset is 750 Hz - fixed, at the moment.)
- The problem with very poor opposite sideband rejection on TX has now been fixed!

* * *

Changes in detail:

Disruption when the STEP+ and/or STEP- buttons were pressed:

Previously, pressing the STEP+ and/or STEP- buttons would often (but not always) cause a "crash", squeak or other disruption, and cause the transceiver to briefly go into TX mode. The problem appears to be due to an undocumented fault with the MPU itself in which slow rise-time on some I/O pins can cause an unrelated EXTIO interrupt to occur.

Within this interrupt, the intended source of the interrupt (which was supposed to be PTT) was explicitly tested to prevent this problem from happening.

Minor bug in the Volume control:

Volume settings 10 and 11 were the same - no longer. Volume settings 11 and up are now one step louder than before, so be careful!

RX Phase adjustment:

The RX I/Q phase may now be adjusted and the setting saved, the value being adjustable from -32 to +32, representing +/- 0.5 degrees of phase adjustment.

To adjust this, tune in a clean, CW carrier on SSB for a 1000 Hz tone, then switch to the opposite sideband. FIRST adjust the RXIQ GN (gain) for minimum signal, THEN adjust the RXIQ PH (phase) to achieve a better null - and then do it again until the best null is achieved. You will note that the phase adjustment is less sensitive than the gain: This is normal.

Due to the nature of a "baseband" (low audio to 10 kHz or so) 0-90 degree Hilbert transformation filter, the amount of opposite sideband suppression will vary in the first few hundred Hz, and it is not possible to get a good null at, say, 300 Hz AND at 1 kHz: A null around 1000 Hz is suggested as that puts the adjusted null largely within the range of the low audio range where it would most likely be noticed from a station on the opposite sideband - but you can adjust it to your taste.

Comment: The scaling of the RXIQ GN (gain) has been changed so your previous setting will have to be doubled - if it was 10, it will now need to be 20. This was done because it was found that with the added phase adjustment, more granularity was needed.

TX Phase adjustment:

The TX I/Q phase may now be adjusted and the setting saved, the value being adjustable from -32 to +32, representing +/- 0.5 degrees of phase adjustment.

IMPORTANT NOTE:

TX IQ Phase may be adjusted ONLY while in transmitting in an SSB mode, because that's the ONLY time that the signal path goes through components that might require phase adjustment: You will find the adjustment of the TX IQ phase (TXIQ PH) grayed out and disabled under any other conditions.

Note that with the improved TX opposite sideband suppresssion (see below) you probably won't need to adjust this, but you will find the procedure below:

- Using a computer program such as Audacity or using an audio tone generator, produce a 750 Hz tone, feeding it to the LINE input of the transceiver and make sure that it is in LINE input (rather than MIC) mode and connect it to a dummy load capable of handling at least 10 watts. Set the mcHF transceiver to USB mode.

- Using another computer connected to another receiver, tune to the transceiver frequency - in USB mode - using a program that shows a waterfall display: Any digi-mode program will do, like Ham Radio Deluxe, DigiPan, MultiPSK.

- Couple a wire "near" the transceiver so that it will get a good signal when you key the transceiver.

- Push the TUNE button on the transceiver to enable transmit and adjust the tuning on your monitoring receiver so that it is centered in the waterfall display. You should see at least two traces: A strong one, and then a weaker one about 750 Hz lower in frequency. Place this weaker one in the center of your waterfall display.

- Adjust the TXIQ GN (gain) control and you should see a third signal appear, 750 Hz below the weaker one: This is the undesired signal that you are trying to null out. Adjust the TXIQ GN until this gets as weak as it will get.

- Press the TUNE button again to go back to receive. Now, using a microphone or Morse key, activate the PTT line with the radio in USB mode.

- If you are inputting a 750 Hz tone in the LINE input - and have it properly selected - you should see a display much like you saw in TUNE mode. If you don't see such a display, make sure that you are generating a 750 Hz tone, putting it into the LINE input at a reasonable level. If you see a lot of noise, reduce the tone level.

- Once you have a reasonable display similar to the one obtained in TUNE mode, you may now adjust the TXIQ PH (phase) control to minimize the level of the "unwanted" signal in the way that was done in TUNE mode.

- It is recommended that you write down your settings. Press TUNE again to return to RX mode - and save your settings - when you are done.

Due to the nature of a "baseband" (low audio to 10 kHz or so) 0-90 degree Hilbert transformation filter, the amount of opposite sideband suppression will vary in the first few hundred Hz, and it is not possible to get a good null at, say, 300 Hz AND at 1 kHz: A null around 750 Hz is suggested as that puts the adjusted null largely within the range of the low audio range where it would most likely be noticed from a station on the opposite sideband - but you can adjust it to your taste

Again, if you do not have the ability to do the above procedure, simply set the TXIQ PH (phase) to ZERO and do only the first part of the adjustment using the TXIQ GN (gain) in the TUNE mode and you should get quite reasonable opposite sideband suppression.

The CW TX and RX frequencies are now aligned with each other, and they fall within the center of the 300Hz and 500 Hz wide CW filters:

Note: The CW demodulation operates using USB.

The CW TX, RX and sidetone frequencies now coincide with each other, and with the centers of the 300Hz and 500 Hz filters. Before, the offsets were 500 Hz and the "zero beat" RX and TX frequencies didn't exactly align - which meant that if you tuned into a station so that its beat note was the same as your sidetone, you would not be on the same frequency as that other station. This has been fixed, with the offset now being 750 Hz.

Note that the LO of the transceiver does NOT shift when going between SSB and CW mode in receive: The displayed frequency is that MINUS the pitch of the tone that you are hearing. In other words, if you were to zero-beat the station you were copying, the displayed frequency would be the transmitting frequency of that other station.

Problem with poor opposite sideband suppression has been fixed:

While in the process of adding the TX phase adjustment, it was noticed that the TX opposite sideband suppression was very poor at low-medium speech frequencies. This was traced to the use of the wrong Hilbert Filter in the TX SSB audio path: The filter that had been chosen was appropriate for up-converted modulation (e.g. SSB generated at, say, 5-20 kHz) and then up-converted to RF, but not for "baseband" audio that started at 300 Hz.

Because a +/- 45 degree Hilbert was chosen, this filter could not have good "near end" (e.g. near 0 or Nyquist) phase response - particularly with a limited number of taps.

Fortunately, another Transform filter - the same one that I'd reworked from the receiver - was already available as it uses a 0-90 degree "phase-added" approach which, with care, can be made to have usable response with a reasonable number of taps, allowing it to be "dropped" into place. Since it is the same transform filter as in receive, it has the same opposite-sideband rejection on transmit as it does on receive.

As described above, one can tweak the nature of the opposite sideband suppression with both the gain and phase adjustments - particularly in the "close-in" area below 500 Hz or so where the Hilbert transform's response starts to get a bit "iffy".

Changes with version 0.0.185

- An "Auto" RF gain setting.
- The Spectrum Scope has been further modified to make it faster and more responsive
- The center frequency appears below the Spectrum Scope, as do abbreviated frequency indicators under the graticules, that move when tuning frequency is changed.
- The bug that had sometimes caused the "White Screen" problem on power-up is believed be fixed.
- The voltmeter has been thoroughly debugged so that it now reads properly at all voltage.
- The voltmeter now turns red at voltages <9.50 volts to indicate a marginal power source.

- The TX IQ Gain adjustment now works.

- The virtual EEPROM can now been written to and read from reliably!

- Operational values, including current frequency, band and mode are now saved on power down/up.

* * *

In detail:

Auto RF Gain setting:

- On the RF Gain setting, there are settings 0-8 as before, but now, in lieu of "9" there is an "AUT" (for AUTO) setting. In this mode the RF gain is incrementally increased when A/D clipping is detected (e.g. the white portion of the S-meter goes red) and these are slowly relaxed only when the signals have decreased for a while. This should allow operation on a crowded band with strong stations without too much problem of A/D overload.

Spectrum Scope modifications:

There have been more extensive modification of the spectrum scope. Its display is still relative to the strongest signal within the +/- 12 kHz display range, and it is possible, under very "quiet" band conditions, for the display to go nearly "white" because of this as its AGC causes its baseline to rise upwards.

PLEASE NOTE: The spectrum scope's AGC is fairly slow, so it takes several seconds to adapt if an antenna is connected or removed!

You will also note that below the spectrum scope, aligned (mostly) with the graticules, you will see the center frequency, rounded to the nearest 1 kHz (which is why it changes at xxx.5 kHz) plus the 10's and 1's of kHz every 6 kHz on either side of the center.

LCD Initialization modification to prevent the "White Screen" on power-up:

It has been noted - and reported - that the LCD would occasionally display white when the transceiver was powered up, although it seemed to work otherwise (e.g. audio, responding to buttons, etc.)

Delays in the LCD initialization routine were doubled: I not sure which one(s) seem to have fixed the problem, but I have not seen it happen since!

Voltmeter debugged, low-voltage warning:

The voltmeter function has been reconfigured so that it should now reliably read voltages now... Really! It does update only every few seconds.

The numbers will also turn red if the voltage displays <9.50 volts, indicating that the power source is likely too low for reliable operation as this will start to reach dropout voltage for the 8 volt supply bus.

TX IQ Gain setting now works:

In addition to the RX IQ gain setting (which works, and the adjustment resolution is now finer), the TX IQ gain setting now works and you can use this to better-null the opposite sideband.

One method to do this is use the TUNE mode and press CAL while monitoring on another receiver set to USB showing a waterfall display. Tune in the main signal on the waterfall at 2300 Hz: Approximately 1000 Hz below the main signal (if you tuned it in using USB on this other receiver!) at around 300 Hz you will find a weaker signal that would be nulled using the TX IQ Gain adjustment.

Please note: While you can adjust the TX IQ Phase setting (and the RX IQ Phase setting as well) these currently do nothing!

The Virtual EEPROM now works!

I have finally gotten the Virtual EEPROM to work reliably: You can look at the code, yourself, to see what was done - although I'm not entirely sure why it made the difference. It would appear that the problem, all along, was reading from the virtual EEPROM.

Operational values are now subject to sanity checks:

With the Virtual EEPROM now working, extensive checks were now implemented to provide "sanity checks" of all parameters that were saved in the EEPROM, initializing them to reasonable defaults should they occur - very important on an EEPROM which could contain... anything! Additionally, all user-interface functions that allowed the user to change parameters were rewritten to properly trap (and fix!) invalid values.

Values saved to EEPROM:

The last-used frequency, band and mode are saved when you turn off the radio using the POWER button: If you simply disconnect the power, don't count on your frequency/band/mode being saved!

At some point in the near future there will be saved the Frequency and Mode on a per-band basis, as well as the last-used filter for the SSB and CW modes.

Now that the EEPROM is working it is now possible to work on the User Interface and add things like:

- VFO A/B
- Memory
- Split

(and many more!)

* * *

Amongst the values saved to EEPROM include:

- PA Bias
- Mic Boost setting
- TX IQ Gain Balance
- TX IQ Phase Balance (saved, but not implemented)
- RX IQ Gain Balance
- RX IQ Phase Balanced (saved, but not implemented)
- TX Audio Source (Mic/Line)
- TCXO State (on/off)
- Audio Gain (e.g. Mic. Boost)
- RF Gain (0-8/Auto)
- RIT
- RF ATTEN (deprecated and will be removed. It still supported in this code, but not for much longer: With the RF Gain code mentioned on the previous release, it is obsolete and Q2 should be removed!)
- TX power level
- Keyer Speed
- Keyer Mode
- Sidetone level

Changes with version 0.0.183

- The Spectrum Scope has been completely rewritten and should be more informative.
- The RF gain control has been implemented using different hardware, eliminating the need for Q2 entirely.
- The S-meter now indicates conventional units (6dB per S-unit, 10 dB thereafter)
- Receiver overload (A/D converter clip indicator) implemented
- Bug in display of "Cal" menu for RX settings now fixed.
- AGC has been improved.

* * *

Spectrum Scope:

- This now uses an IIR averaging filter and nonlinear display so that greater dynamics can be displayed. The averaging effectively improves the resolution and increases the persistence of the display.
- For step sizes 1 kHz and smaller, the display is not redrawn, but for step sizes larger than 1 kHz (e.g. 10kHz and up) the averaging is cleared on each step.

When signal level change radically, it may take a moment for the spectrum scope to recalibrate itself. Also, with very low signal levels (e.g. no antenna or a dead band on a very quiet antenna) you will probably see a semi-random assortment of lines.

* * *

RF Gain control:

The RF gain control is implemented using the hardware gain control of U1, the WM8731 codec. This significantly outperforms the capability of almost any practical front-end attenuator and, certainly, Q2, allowing the receiver to tolerate signals as high as -25dBm on 40 meters - the actual limit of the hardware.

The RF gain control still goes from 0 to 8 with 0 being lowest gain - and a fairly deaf receiver - to 8, the "normal" setting of previous firmware, which had been always set at "maximum" gain.

This control is helpful to prevent receiver overload - ***see the next section.***

I will probably remove the on-screen "ATT" control soon and use it for something else - probably for threshold adjustment of the experimental noise blanker.

* * *

S-meter recalibrated:

The S-meter is now calibrated to read according to the "standard" convention of:

- 6 dB per S-unit
- 10 dB per 10dB division above S-9
- 50uV (-73 dBm, 50 ohm) calibration at S-9 (with Q1 fitted)

The S-meter is now keyed from the AGC and it is also calibrated against the setting of the RFG control: As the gain of the receiver is reduced, the gain calculation of the S-meter is compensated so that a given signal level will remain constant ***as long as it is strong enough to be detected at the reduced gain level.***

IMPORTANT NOTE - Please read this and understand:

Because of the 50uV calibration at S9 and the 6dB per S-unit, you will note that you will NOT be able to get an "S-0" meter reading - at least with this version of firmware.

You must expect to see an S-3 to S7 reading from the background noise of the receiver - less if you have modified it and added some metal shielding between the UI and RF boards. You should consider this to be a tool when you work to reduce the various noises emanating from your receiver.

* * *

Receiver overload indicator:

Previously, the A/D converter was set to maximum gain, causing the receiver to be susceptible to overload at signal levels on the order of -45 to -50 dBm when preamplifier Q1 was fitted - but it was not necessarily easy to see if it was overloading, particularly if a signal doing this was outside the range of the spectrum scope.

Now, the lower portion of the S-meter (from S-1 to S-9) will turn red if the receiver's codec is overloading. If this happens, reduce the RF gain with the RF gain control to the point where it no longer flashes red.

If the receiver overload indicator flashes occasionally - on static crashes - there is probably no need for concern, but if it continually flashes, you should turn the RFGain control down a bit to avoid splattering - although you may not necessarily hear any degradation.

One of the first signs of overload is that the spectrum scope's "noise floor" will rise up. While this will happen with static crashes, too, you will soon learn to recognize when this happens because of nearby, strong signals.

* * *

RX Calibration menu display bug fixed:

This was a bug where the RX calibration menu would not always disappear properly when it was invoked.

AGC improvement:

There were some minor issues with steady-state signals on the AGC that have been fixed. The support of the Q2 RF attenuation has been removed as the addition of the manual RF gain control described above is superior. Unfortunately, using the above attenuator as part of the AGC is not practical as changing the gain settings using that hardware causes audible "clicks" to occur and test code of the AGC with it resulted in a quiet - but audible - "motorboating" as it was changed.

* * *

Outstanding issues:

- Saving data in EEPROM: This is being investigated as to why this does not work. While the adjustment of the RX phase *does* work, I would recommend leaving it at ZERO for the time being, until the saving/retrieval of that data is debugged or else you will probably get strange values (e.g. 65530 or some other large number.) I've not really spent much time in chasing down this problem, however.

- A "white screen" when powering up. This is a bug that was inherited from previous firmware and has to do with the hardware not initializing properly. If this happens to you, remove the DC power for 5 seconds and then try powering up again.

- Various beeps and clicks when going between TX and RX. Again, I have not attempted to address this issue.

Changes with version version “0.0.181_1”

1 - An AGC algorithm has been implemented. It is currently "fixed" at a "medium" decay speed. **PLEASE read the note below about increased noise and feedback if you use this firmware with AGC!**
2 - The CAL function has been modified so that there is now a gain adjustment for receive I/Q to better-balance the receiver.
3 - The tuning no longer truncates the numbers to the right of the tuning step when you increase the step size. T
4 - The spectrum scope has been de-linearized somewhat to have slightly better dynamics display on the screen.
5 - The spurious "1" that appears on the voltmeter if you power up the radio on a power supply of less than 10 volts has been fixed. (e.g. running from a 9 volt supply would display as "19.00V")

Of these, the first two require some explanation:

* * *

AGC:

There is not (yet) any menu setting for this, so the AGC is "fixed" at what sounds like a fairly good compromise set of time constants for both SSB and CW.

This AGC operates with three separate mechanisms:

1 - Via a post-filter gain control for weak/medium-strength signals.
2 - For medium-strong signals, the bias of Q2 is increased as the signal level increased to prevent overloading of the A/D converter. If Q2 is not installed, this will have no effect. This can extend the strong signal tolerance of the receiver. Because of this, the "ATT" control is no longer functional in any real way at the moment, but this could be fixed in a later version of code.
3 - For somewhat stronger signals, the pre-DSP gain is reduced (although this needs to be revised to use the CODEC's hardware gain control). The is the exact same mechanism as the "RFG" control.

The RFG control now acts in way similar to that of the "RF Gain" control on many radios in that setting this lower will reduce the level that AGC will be allowed to increase the gain of this third mechanism.

Bugs:

- There is an interaction between #2 and #3 with very strong, steady-state signals that needs to be resolved that can manifest itself as a slight, slow clicking. This only occurs with *extremely* strong signals, however.

- As the DAC that adjusts Q2 adjusts voltage, there can be a slight clicking as the D/A steps through its settings. This could probably be fixed with an additional resistor and capacitor, but if you don't have Q2 installed, you won't hear it at all - and you probably wouldn't ever hear it in normal operation, anyway.

- You WILL hear more receiver noise! Because this is an AGC, it will cause the audio gain to be cranked up and the odd noises that were there before will be even more obvious now!

- You may get speaker feedback. I had to change my 2.7 ohm resistor on the power supply line to U2 (the speaker amplifier) to 4.7 ohms. It is also recommended that the value of C32 be increased with the largest value capacitor that will fit within that footprint.

- If you didn't have a problem with TX audio getting back into the TX mixer and causing feedback before, you will likely have that problem now with the increased system gain! Please read the recent postings on how to solve that problem.

Note: There is an unrelated bug in the volume control (AFG) in that if you turn it down below 10 - and turn it back up, the volume will not actually return to its previous setting. This bug was present before, and I've not had time to chase it down. You can work around this pressing either of the STEP buttons (press the STEP+ then the STEP-) a time or two which often resets the CODEC.

To be added:

- Menu-selectable modes for slow, medium and fast AGC action. The User Interface will need to be modified in some way for this - but I'm not sure how...

- Non-linear AGC decay to compensate for the fact that signals are logarithmic, but the gain values within the AGC control loop are not! This is not really noticeable to the user unless he/she carefully observes the reaction of the AGC's dynamics.

Notes related to older firmware releases by Chris, M0NKA:

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mcHF v 0.0.0.181 - 2 June 2014

- Improvements on the TX SSB modulator DSP routines, now there is low pass audio filter before the tx modulator

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mcHF v 0.0.0.171 - 4 May 2014

- New entry into power level menu - 'Full', it will not limit the output power level
- Bug fix, plugging/unplugging a key or microphone during bootup was causing stall of the firmware
- CW Keyer mode, speed and sidetone volume values will now be saved to the virtual eeprom

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mcHF v 0.0.0.170 - 28 April 2014

- Updated power coefficients for 20m, it seems they were wrong due to badly adjusted LPF
- bug fix when loaded band from eeprom was not matching saved frequency
- added 60m band

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mcHF v 0.0.0.169 - 24 April 2014

- Updated LCD driver, now HY28A with parallel and HY28B with dual interface are supported. Autodetect is used, so all three LCD are plug and play, no need to cut traces on the board. The HY28A serial version talks to the CPU via SPI, where the HY28A parallel version and HY28B dual version talk via 16bit parallel interface

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mcHF v 0.0.0.168 - 15 April 2014

- SWR protection implemented, it will be activated on too much reflected power, PROT indicator in red will show. To reset, change band or restart
- Bug fix, FWD and REF measurements swapped in software
- Updated TX power output levels for the indicator
- Fixed power factor values for more even distribution in different bands
- Activated the power level control, now will affect the output level in Tune, CW and SSB modes

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mcHF v 0.0.0.165 - 5 March 2014

- The I2C bus speed for SI570/MSP9801 increased to 100 kHz
- CAT control driver implemented - for now only PTT works and not possible to stop the driver(beta testing phase)

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mcHF v 0.0.0.164 - 3 March 2014

- Critical bug fix related to the setting of the SI570 frequency

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mcHF v 0.0.0.163 - 2 March 2014

- Virtual Eeeprom implemented, now all tunning values will be saved and on power restart last transceiver state will be restored

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mcHF v 0.0.0.162 - 28 Feb 2014

- Image relocated at 0x08010000 and called from device bootloader at default reset vector
- Update is no longer via dfu image, but flat bin file and the USB utility is to be used for that

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mcHF v 0.0.0.161 - 6 Feb 2014

- Reworked UI driver, now have better consistency of the controls on screen
- Calibration mode implemented, PA BIAS is the only working menu for now

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mcHF v 0.0.0.160 - 16 Jan 2014

- fix of excessive software volume
- temporary fix for SI570 unreachable frequenciers - the segments will be painted red to show refusal by the LO to accept new freq
- restored TRX4M mute during freq change, to reduce speaker clicks

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mcHF v 0.0.0.158 - 11 Jan 2014

- Better display of the noise floor in the Spectrum Display indicator

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mcHF v 0.0.0.157 - 10 Jan 2014

- Soft tcxo routine, change of si570 freq only if change of 1 degree detected to reduce too much activity on the I2C bus
- Bug fix from previous build when tcxo will not lock after off/on sequence

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mcHF v 0.0.0.155 - 5 Jan 2014

- First firmware release