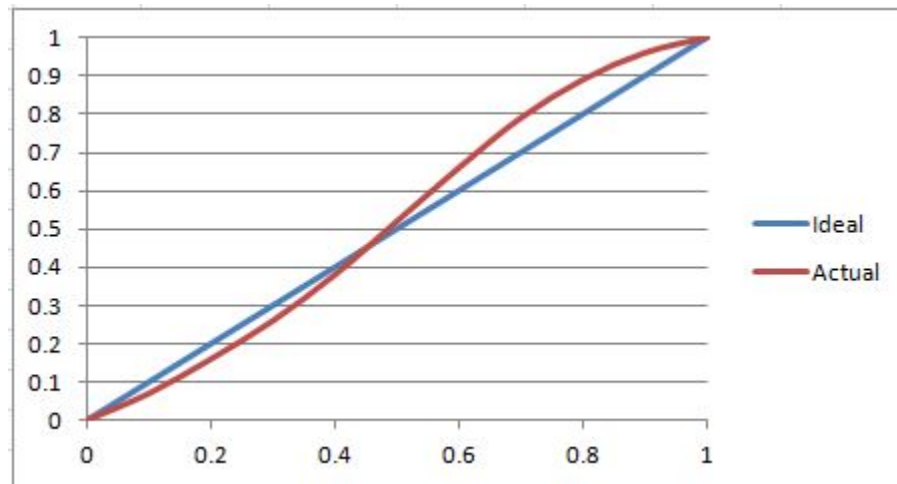


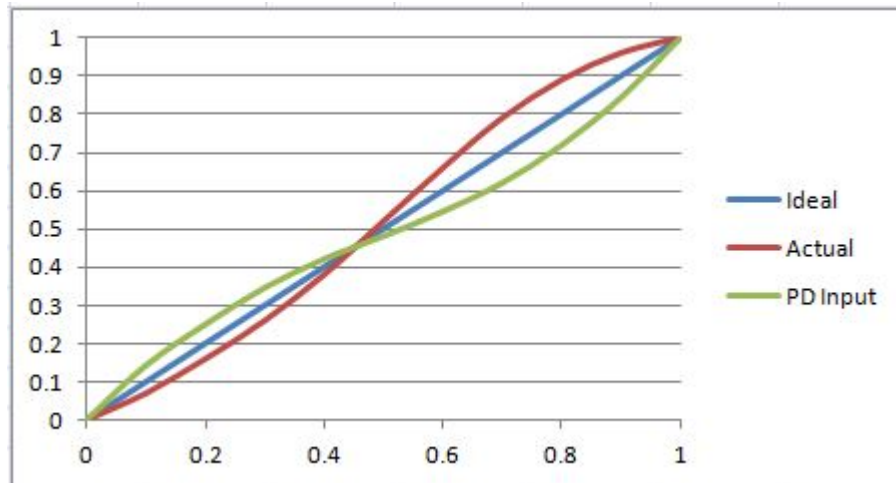
Introduction

Linearity of an amplifier implies that the output increases linearly with increases of the input signal. I.e., if an input of 5W produces an output of 50W, then an input of 10W should produce an output of 100W. Unfortunately, the analog amplifiers in our power amplifier stages are not so perfect. A real amplifier might behave more like the figure below.



The unfortunate side-effect of this NON-linearity is Inter-Modulation Distortion (IMD), some of which results in “splatter.” Consequently, on the panadapters of our SDRs it’s easy to see signals that extend outside a normal transmit bandwidth. This phenomenon is more evident on strong signals as the IMD may be 25dB or more down from the peak of the signal. This IMD pollutes our bands and raises the interference and noise levels for amateurs that may be operating many kilohertz away from the offending transmitters. Some of the IMD is less visible on panadapters because it falls IN-BAND, i.e., within the bandwidth of the transmitted signal. This adds distortion to the signal.

Fortunately, several techniques have been developed to improve the linearity of analog amplifiers and thereby reduce the IMD levels that they generate. One such method is “Predistortion”—making the amplifier’s output more linear by “predistorting” the input signal in such a way as to offset the distortion that the amplifier will produce. The output of the amplifier can therefore be corrected to behave very much like the ideal case shown above. (See figure below.)



In addition to the amplitude distortion shown in the above figures, there is also a phase distortion (amplifier phase shift varies as a function of signal amplitude) that must be corrected to achieve significant reductions in IMD.

The computing power in our SDRs can be used to calculate the required corrections and then to apply those corrections to the digital transmit samples. However, to be able to calculate the corrections, it is necessary to know the amplitude and phase characteristics of the amplifier to be corrected. Each amplifier is slightly different and its characteristics also vary with frequency, temperature, load impedance, and supply voltage. It follows that, for best results, we must have a method to measure the characteristics of the amplifier while it is in use and to “adapt” the correction as conditions change. This approach, used in PureSignal, is often called “Adaptive Predistortion.”

We can measure the characteristics of the amplifier by using the receiver of our SDR to receive the amplifier’s output and then comparing the signals going into and coming out of the amplifier. Note, however, that receiving the amplifier’s output signal requires a feedback path to be established from the amplifier output to the receiver input during the time that we are transmitting. Clearly this must be done with a directional coupler/attenuator or sampler/attenuator so as to deliver ONLY the required amount of power back into the receiver input.

Hardware Configuration

Feedback

As mentioned above, calibration requires that the amplifier output be fed-back to the receiver during transmit. Generally, this will involve using one or more relays such that the receiver is (1) connected to the antenna during receive, and (2) connected to AN APPROPRIATELY ATTENUATED sample of the transmitter output during transmit.

- ***YOU ARE RESPONSIBLE TO ENSURE THAT YOUR FEEDBACK IS ALWAYS SUFFICIENTLY ATTENUATED SUCH THAT IT DOES NOT DAMAGE YOUR RECEIVER HARDWARE.***
- For proper operation, feedback MUST NOT create an ADC Overload. For PowerSDR_mRX, ADC Overload will be reported via a flashing warning across the bottom of the panadapter.
- For accurate calibration, the feedback level must be as close as practical to ADC Overload without actually causing an overload. A green-yellow-red indicator is provided on the "Linearity" form to report the feedback level. (See "PureSignal Controls & Operation" below.)
- With a radio attenuator setting of 0dB, ADC Overload occurs at approximately -11dBm. For best results, feedback should be above -17dBm and, as stated above, must NOT create ADC Overload. With a higher radio attenuator setting, you can run a higher feedback level into the receiver input.
- Helmut, DC6NY, has provided an excellent design for an RF Sampler that can be used to capture the feedback signal.
- See: < http://www.hamsdr.com/data/GlobalFileUploads/9636_RF%20Sampler%20for%20Pre.pdf > , RF Sampler for Pre-Distortion Loop ".

Alex Feedback

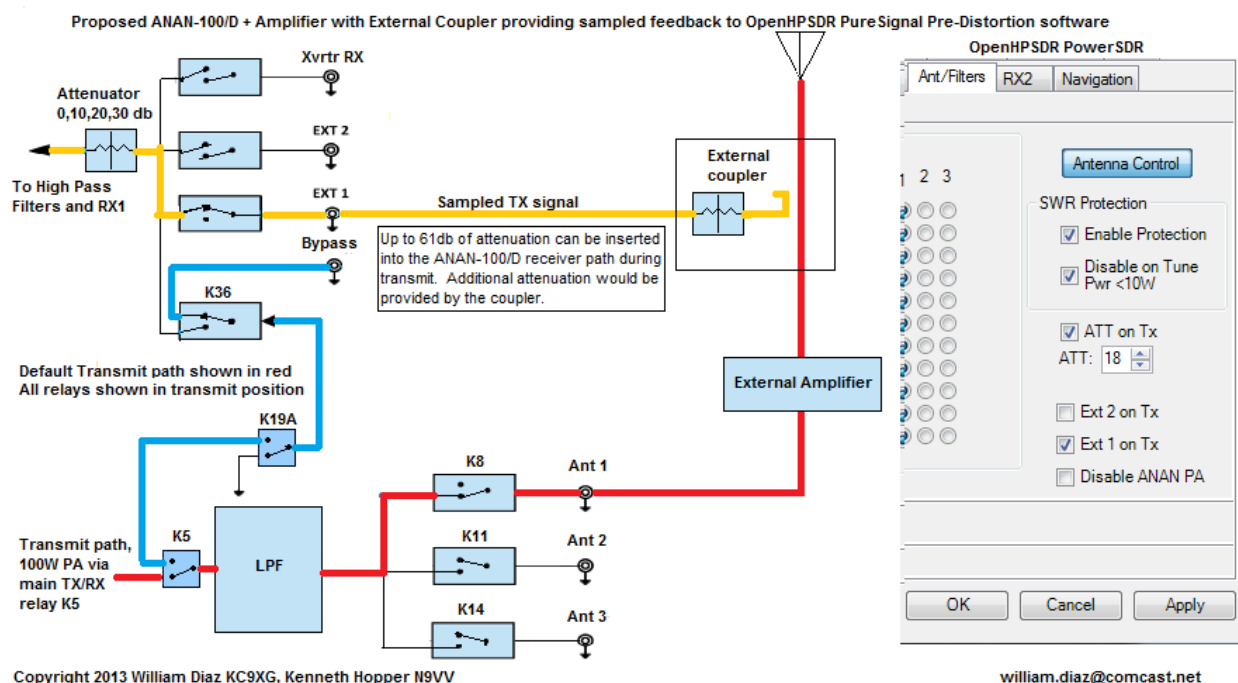
For systems using the Alex filters, you have two options for PureSignal feedback. On the Setup > General > Alex > Antenna tab, check boxes "RX1 IN on TX" and "RX2 IN on TX" are provided. If one of these is checked, during transmit, other settings will be over-ridden and the chosen RX connector will be connected to the receiver input filters. Therefore, you can input your feedback to the chosen RX connector. Alternatively, if you provide your own T/R switching (such as using the relay on the coupler designed by Helmut, DC6NY, and referenced above), you can insert feedback between the "Out to Rx" connector and your receiver card input.

ANAN-100/D Feedback

For the ANAN-100/D, on the Setup > General > Ant/Filters > Antenna tab, two check boxes, "Ext 2 on Tx" and "Ext 1 on Tx" are provided. If one of these is checked, during transmit, other settings will be over-ridden and the chosen EXT connector will be connected to the receiver input. Therefore, a coupler providing feedback can be connected to the chosen EXT connector.

For the ANAN-100/D, testers have noted that there is a very high level of crosstalk between the transmitter output and receiver input even without a direct feedback connection. That feedback MAY be sufficient for accurate calibration of a "barefoot" ANAN-100/D. One disadvantage of that approach is that this feedback is due to stray coupling and the feedback level is likely to vary substantially from band to band. Using this stray coupling will NOT work to linearize a following linear amplifier since "clean" feedback from the output of the amplifier must be provided.

The following diagram illustrates the ANAN-100/D with an external amplifier and coupler.



Amplifiers

The current PureSignal algorithm does not attempt to dynamically compensate for “memory effects.” Memory Effects are phenomena whereby the amplifier gain and phase are not only a function of the current input signal, they are also a function of past input signals. The amplifier “remembers” signals that went through in the past. These effects can be thermal or bias / power-supply related. For example, suppose that a strong signal over a period of some milliseconds heated the active devices of the amplifier thereby changing their gain and other characteristics. Then the amplifier “remembers” those characteristics until it cools back down even though a weaker signal may pass through in the meantime. So, for the same weak signal level, the amplifier characteristic is different depending upon whether it is hot or cooler, depending upon the preceding signals. A similar situation exists with bias and supply voltages. Suppose a strong signal momentarily causes a slight drop in power supply voltage. If the supply does not recover instantaneously (which it typically will not) the amplifiers gain will have been changed for a subsequent weak signal.

Amplifiers with weak memory effects will be more effectively corrected than those with strong memory effects. Among solid-state amplifiers, current experience suggests that LDMOS amplifiers using higher voltage (~50V) supplies produce the best results. In any particular amplifier design, factors that can minimize memory effects include:

- High-voltage / low-current designs to minimize power supply droop.
- “Stiff” power sources to minimize voltage “sag.”
- Lower capacitance devices (such as LDMOS).
- Excellent heat-sinking, probably including a copper heat-spreader.

As you would suspect from this list, 13.8V FET amplifiers can be challenging to correct. However, stiff power supplies and good cooling can go a long way.

Limited testing of this software has been done with vacuum tube amplifiers. Your results might be less predictable with vacuum tube amplifiers because the delay times through the amplifiers are much longer due to the high-Q tuned circuits.

THE AMOUNT OF IMD CORRECTION YOU ACHIEVE WILL DEPEND UPON THE MEMORY EFFECTS OF YOUR AMPLIFIER AND THE QUALITY OF YOUR FEEDBACK SIGNAL.

Firmware Requirements

Special features are required in the firmware for PureSignal operation.

- Firmware for Hermes/ANAN-100 must be Hermes_v2.5 or later.
- Firmware for Angelia/ANAN-100D must be Angelia_v2.2 or later.
- Firmware is not currently available to enable PureSignal on Atlas-based systems.

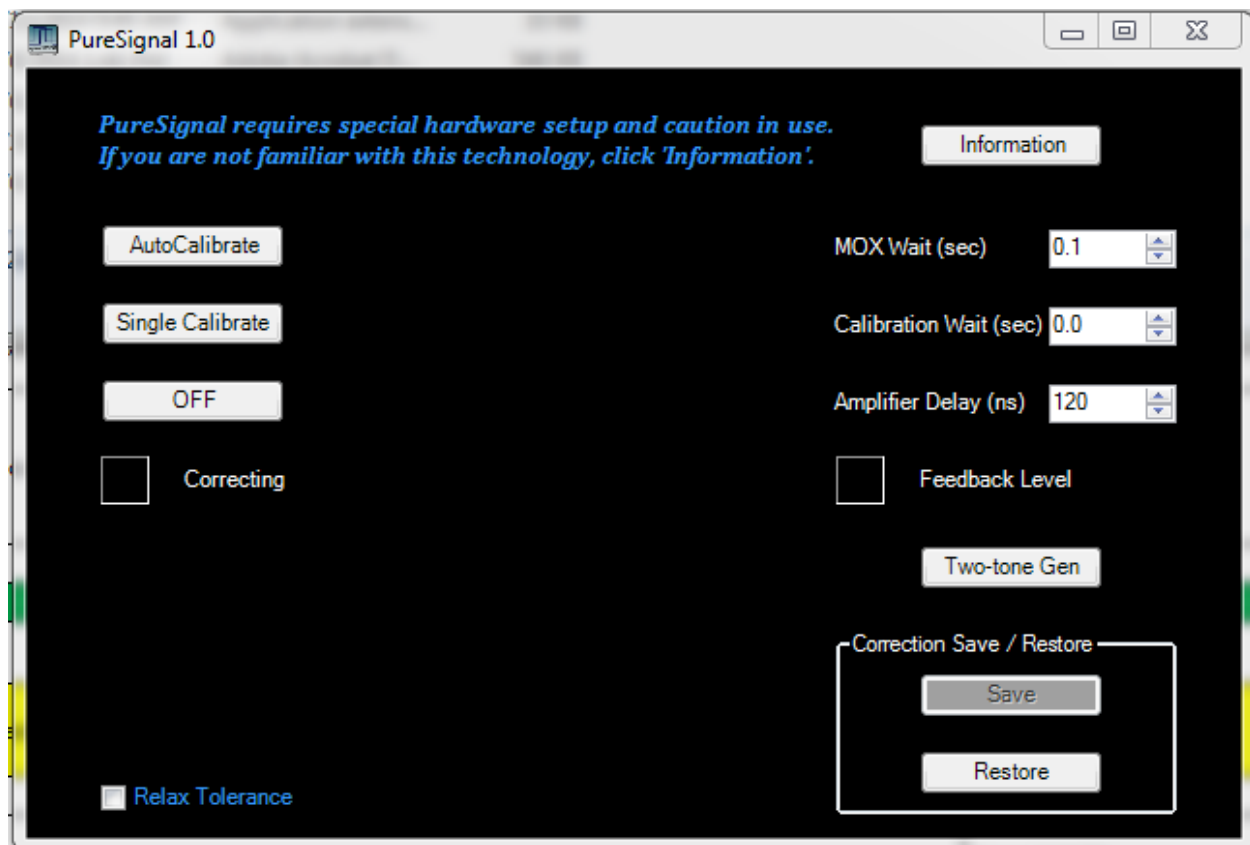
PowerSDR_mRX Settings

- Sample Rate: On the Setup > Audio tab, please set the sample rate to 192K. When Gigabit Ethernet firmware is available, in the future, we will move to 384K.

PureSignal Controls & Operation

First of all, you must be sure the box "Disable PureSignal" is NOT checked on the "Setup>General>Hardware Config" tab. Why is this box even there? Having PureSignal enabled MAY require more network bandwidth. Therefore, for users that are operating wireless or have limited bandwidth, it would be best to check the box unless they want to use PureSignal. This is very similar to the situation with the "Limit Stitched Receivers" checkbox just above. At this point, we have made Stitched Receivers and PureSignal mutually exclusive.

Clicking 'Linearity' on the menu strip along the top of the PowerSDR_mRX console will open the PureSignal form.



Operation of the buttons and indicators is as follows.

- **Information:** Clicking information opens a pdf document giving an overview of linearity, pre-distortion, and how to use PureSignal.
- **AutoCalibrate:** Activates / de-activates automatic calibration. The button background color will be green when automatic calibration is turned ON. Note that merely turning ON AutoCalibrate does not ensure that you are actually calibrating and correcting your signal. More on this later. When automatic calibration is functioning correctly, the software will be repeatedly measuring the amplifier response and adapting the correction to adjust to current operating conditions.
- **Single Calibrate:** There may be instances where you do not wish to repeatedly calibrate; but, you instead wish to just take one set of amplifier measurements, calibrate once based upon those measurements, and continue to correct with that calibration data. Clicking this button initiates such a sequence.
- **OFF:** OFF turns off any current correction and collection of amplifier data. It leaves PureSignal in a 'reset' state. You will need to click AutoCalibrate, Single Calibrate, or Restore a calibration data set to return PureSignal to an operating mode.
- **Correcting:** The "LED" labeled Correcting will be green when correction is enabled and a valid set of calibration data has been computed or loaded. When green, your transmit signal will be corrected using the calibration data.

- **MOX Wait:** After you activate PTT or trip VOX, a small amount of time will be required before signal is actually flowing, relays have engaged, amplifier bias has been enabled, etc. MOX Wait is the amount of time from your initiation of transmit until the software will begin collecting data for a calibration. The default of 0.1 seconds should be more than sufficient for most cases.
- **Calibration Wait:** The amount of time (in seconds) between the completion of computation of a calibration and the beginning of data collection for the next calibration. If set to zero, calibrations will proceed in rapid succession as long as valid sample sets can be collected. Higher numbers will increase the time between calibrations and save a few CPU cycles.
- **Amplifier Delay:** All analog amplifiers introduce some amount of signal delay. For solid-state PAs, the largest component of delay is generally introduced by the low-pass output filters. The algorithm used to compute calibration is reasonably tolerant of delay and to an extent automatically adjusts for it. However, you may find that you get improved results by adjusting this control to manually offset the delay through the amplifier. This offset can be adjusted in 20ns increments.
- **Feedback Level:** This “LED” indicates the strength of the feedback signal (the amplifier output signal that is being received back through your receiver). The indicator will be updated ONLY when a complete set of data has been collected for calibration and will then gradually fade to the background color after its update. Green indicates good feedback; yellow indicates that feedback is lower than required for optimum results; red indicates that the feedback level is very low and erratic or erroneous results are likely to be produced. Note that if a complete set of data is not able to be collected, this indicator will never update; it will remain at the background color.
- **Two-tone Gen:** For convenience, a two-tone generator button is provided here for testing. This button activates the Two-tone generator on the Setup > Tests tab. For use with PureSignal, set the Level to 0.0dB so that PureSignal can collect data across the full output range.
- **Correction Save / Restore:** Anytime a valid set of calibration data is in use, that data can be saved to a file for later retrieval. Clicking Save opens the file save dialog for that process. Clicking Restore opens the file dialog to retrieve a previously stored set of calibration data. Restoring a set of data automatically activates correction.
- **Relax Tolerance:** Checking this box is NOT recommended for typical operation. It relaxes a requirement on consistency of samples collected for a calibration. When might this be useful? If you have an amplifier with severe memory effects for which PureSignal refuses to calibrate, OR, in a situation such as a tube type amplifier with AC ripple on the plate supply that is modulating the output. The risk in checking the box is that it increases the possibility of an invalid calibration that the algorithm believes is valid.

VERY IMPORTANT Notes About Calibration

- Collection of a set of amplifier data for calibration requires measuring the entire amplifier characteristic from 0% to 100% of the input level for a particular “Drive” setting. (This 0% to 100% refers to your digital signal level as measured, for example, by the ALC meter; it does NOT relate to your “Drive” setting.) Your signal MUST traverse that entire range (or very nearly so)

for calibration to proceed. The higher signal levels are where most of the IMD is generated; so, getting to virtually 100% is especially important and cannot be overlooked. Therefore, in normal operation your ALC meter SHOULD be periodically (even fairly frequently) hitting 0dB. There are also other reasons this is good practice, for example, in order to use the DAC's full dynamic range. You will likely want a few dB of Leveler gain and/or a few dB of compression to facilitate this process. Unlike analog and many other digital radios, hitting the maximum (0 dB) does not generate any "flat-topping" or material distortion.

- Automatic Calibration is recommended because it adapts dynamically to changing operating conditions. Should you choose to use the Save / Restore capability, be aware that correction that is Restored must be used under exactly the same conditions as it was Saved to achieve good results. It's fairly obvious that significant changes in frequency, supply voltage, and load impedance are conditions that would invalidate a Saved correction. However, also be aware the any change in the "Drive" setting invalidates the correction.
- Due to memory effects, calibration with a two-tone signal will generally NOT produce an optimum calibration for voice and vice-versa. Again, automatic calibration is recommended to allow for adaptation to changing conditions.

Correction Bandwidth

PureSignal can only *compute correction* for IMD products that it can "see." I.e., it can only compute correction for IMD products within the bandwidth of the receiver used to receive the output of the amplifier. So, for example, at 192K sample rate, and allowing for a little filter roll-off, we have a bandwidth of ~186K, or +/- 93K from the transmit frequency.

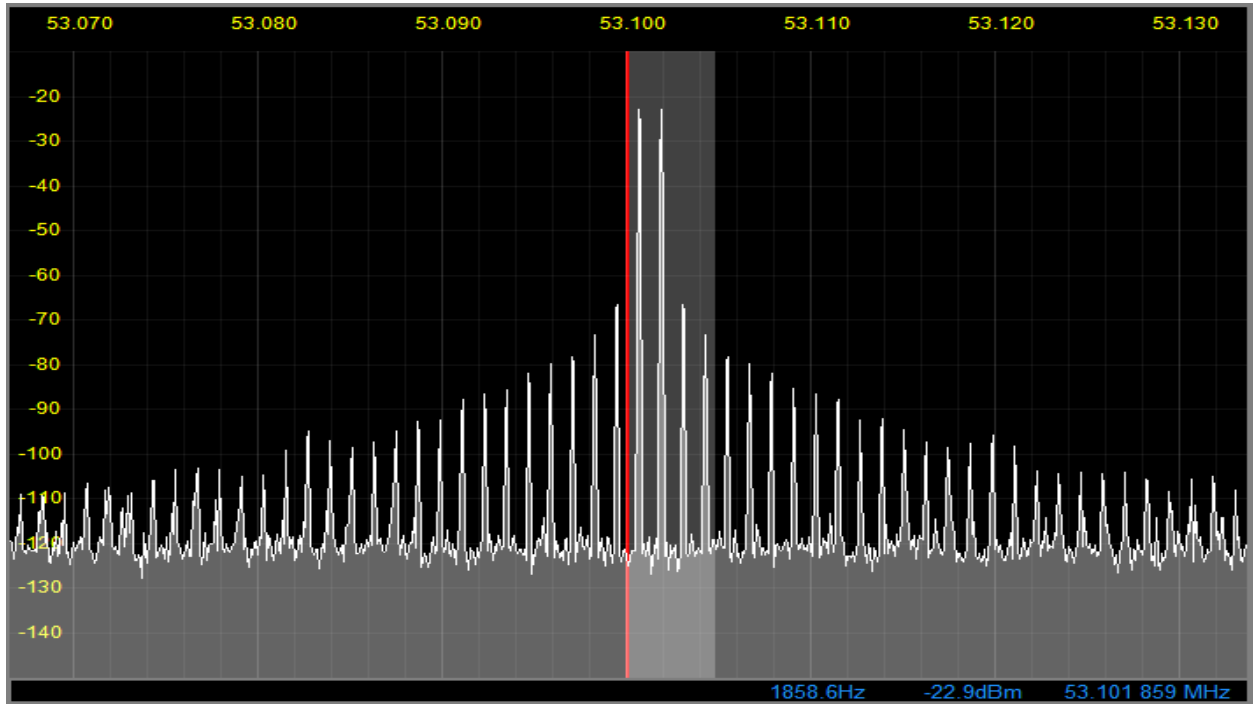
However, PureSignal can only *correct* within the bandwidth of the channel between the correction algorithm and the amplifier to be corrected. We use a 48K sample rate to transfer transmit I/Q samples from the computer to the radio hardware. So, allowing for a little filter roll-off, the correction bandwidth is ~40Khz, or +/- 20Khz from the transmit center frequency.

Unless you have a very dirty amplifier or are significantly overdriving your amplifier, nearly ALL of your IMD energy should fall within this +/- 20Khz bandwidth.

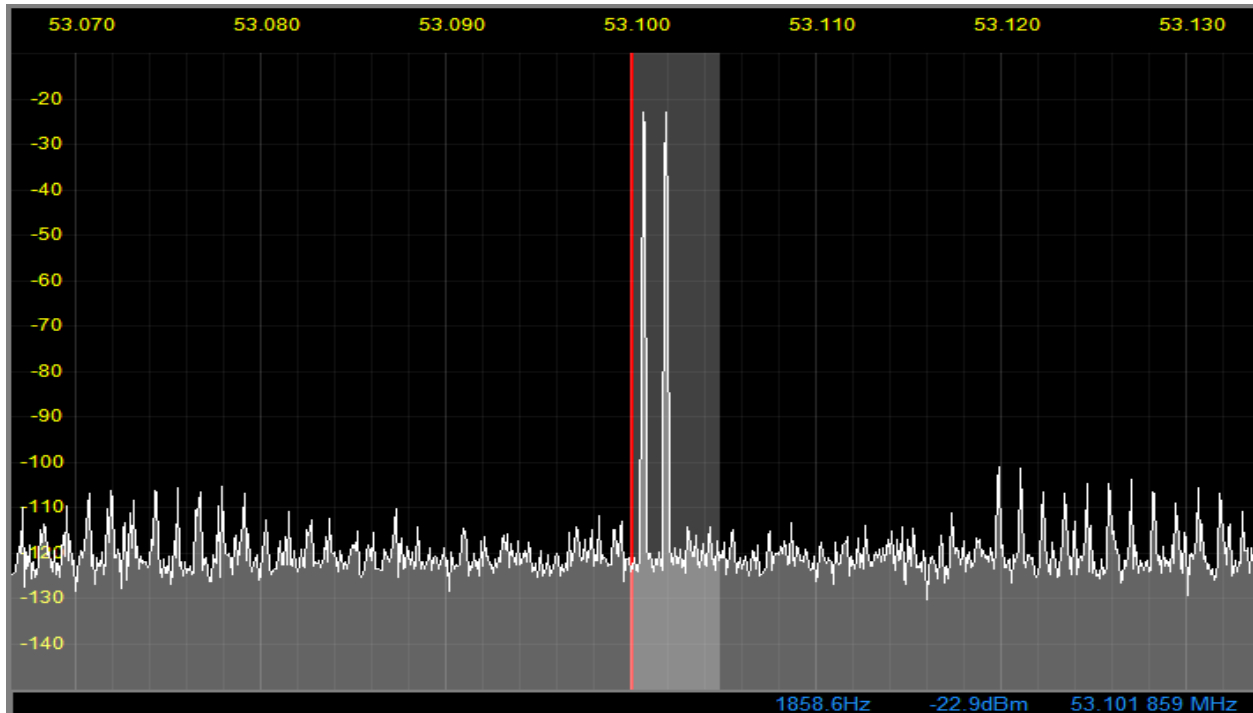
Example Results

Example 1: "barefoot" Hermes card on 6M (results from Warren Pratt, NR0V)

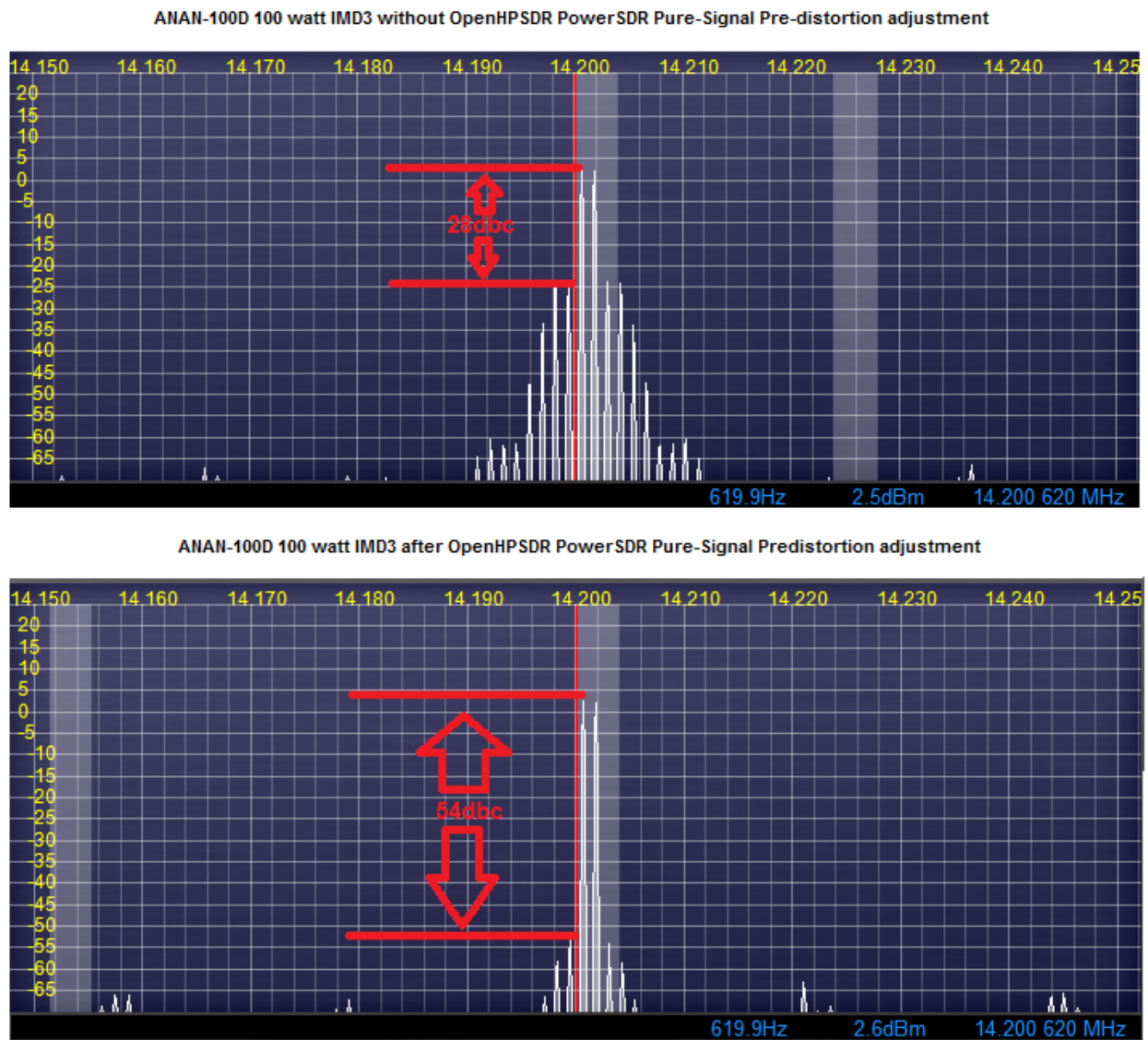
Uncorrected



Corrected



Example 2: ANAN-100D, Stray Coupling Feedback (results from Bill Diaz, KC9XG)



Additional PureSignal Information

See documents posted by Bill Diaz, KC9XG, on the Apache-Labs Yahoo Group.