# Elecraft K3/K3S Noise Reduction

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According to Wikipedia "LMS algorithms are a class of adaptive filters used to mimic a desired filter by finding the filter coefficients that relate to producing the least mean square of the error signal (difference between the desired and the actual signal). It is a stochastic gradient descent method in that the filter is only adapted based on the error at the current time."

For more details go to Wikipedia at, https://en.wikipedia.org/wiki/Least_mean_squares_filter

The K3/K3S employ a similar LMS algorithm in its DSP noise reduction firmware. Unlike most modern transceivers that have NR settings from 0 to 10, The K3/K3S settings are a bit more complex with 24 settings. Settings from 0 to 10 simply adjust how aggressive the NR attacks the noise but Elecraft added delay settings to the NR timing. Other settings mix the noise reduction with the DSP's raw input.

**The instructions (taken from the K3S user manual) are as follows;**

Hold to display the NR setting, which is saved per-mode. Use the VFO B knob to tailor NR for the present band conditions. In general, the higher the number, the more aggressive the noise reduction. Settings **F1-1** through **F4-4** are recommended. **F5-1** through **F8-4** use a different algorithm, where the **-x** part of the setting indicates the degree of mix between the DSP-processed and unprocessed signals (**-1** is about 50% processed, **-4** is 100%). A small **M** appears to remind you that a **M**ixed setting is in effect, e.g. **NRᴍ F5-1**.

Trying to make sense of the various settings, I was baffled as to which one was the best and what are the characteristics of each setting. Working in telecom industry in engineering, I was involved with echo canceller design. Using the LMS algorithm, echo cancellers converge on the sinusoidal elements of human speech and by manipulating the filter coefficients, it can cancel out that annoying echo on a long-distance phone call.

With that experience and to understand the K3/K3S NR, I decided to setup my own test. I wanted to inject a real atmospheric noise source and an adjustable RF signal. My K3S was connected to my HF beam. To inject a low-level RF signal, I connected my HP 8657B RF generator to a dipole antenna that was just below the beam.

Human speech contains sinusoidal elements that the LMS algorithm adapts to and creates a filter around the speech waveform. The waveform constantly changes and to conform to the human speech patterns, the LMS filter must continuously change the filter's shape. Under these conditions, it is too difficult to characterize the K3/K3S NR adjustments. My test setup is a single carrier on 14.200MHz, with my K3S tuned so that the carrier tone is 1.5 KHz, the noise level was S3 and the carrier was S7. To obtain a -15dbm level, I adjusted the K3S line level but it was difficult to get it exactly spot on. An occasional noise burst would bump the level,

The spectrum software was SDR Sharp and was virtually connected to something listed as "I/Q from sound card". It turns out it's the K3S USB audio that was displayed in the software's spectrum. To get the correct frequency span and level, I had to do some fiddling with the SDR Sharp settings. The images on this document was taken from a screen print and cropped using Microsoft paint. The cropped spectrum was saved as a .png file and pasted into this word document.

The purpose of this test is to determine the noise characteristics at each setting and to compare the SSB audio spectrum at 2.8 KHz. During normal QSO's, I notice that there's almost no differences between settings Fn-2 and Fn-3. From Fn-1 to Fn-4, the noise levels dropped quite a bit. On any NR setting SSB voice sounded as if a treble control was turned down. Regardless of the NR settings, the carrier level in the following screens appears to be at a constant level. Using multiple tones, in the future, I plan to do more testing.
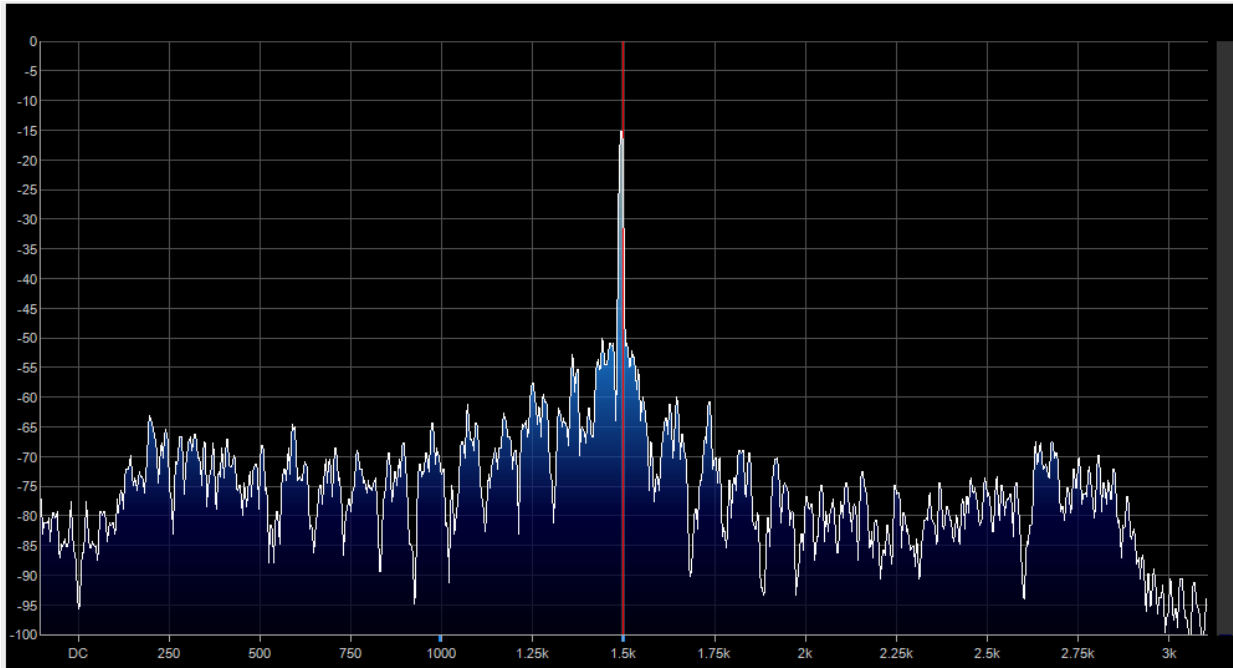
NR turned off, initial settings. Average noise floor is about -55dbm
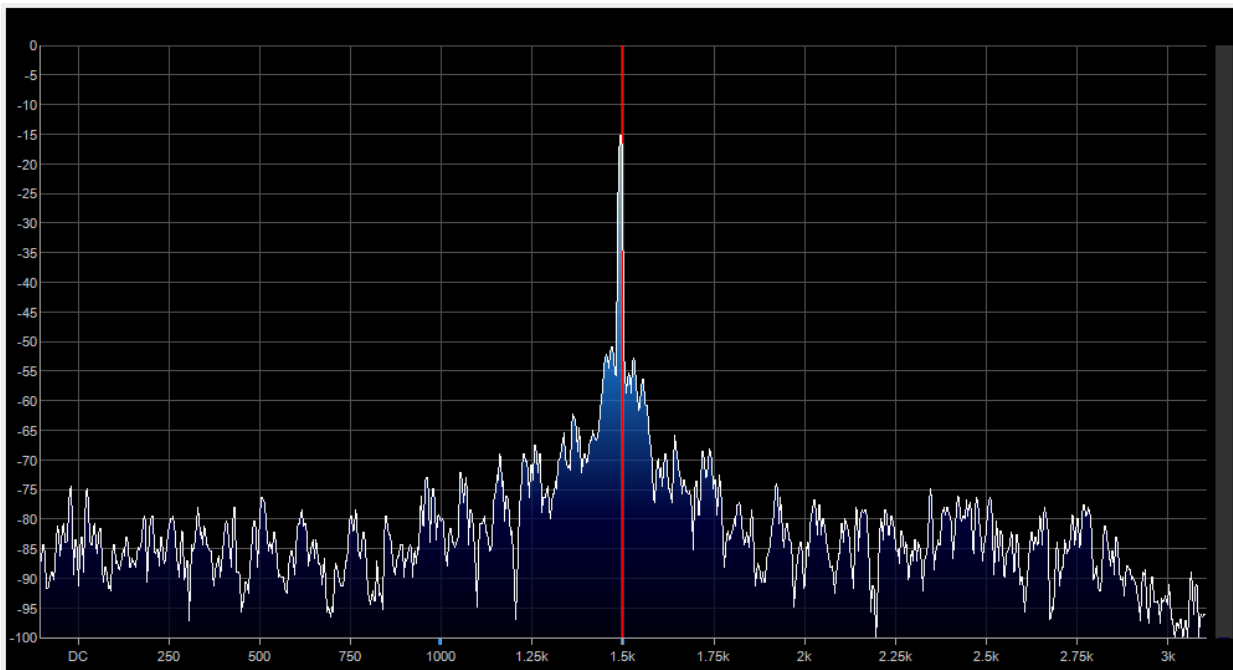
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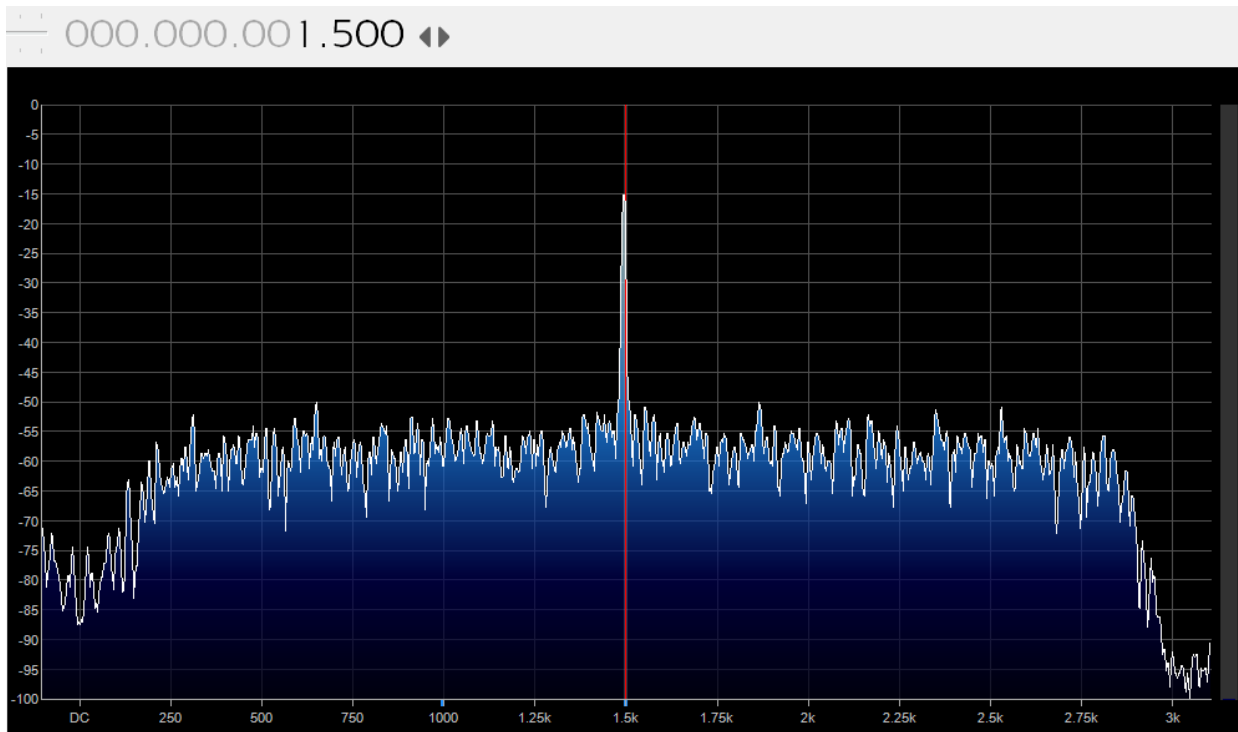


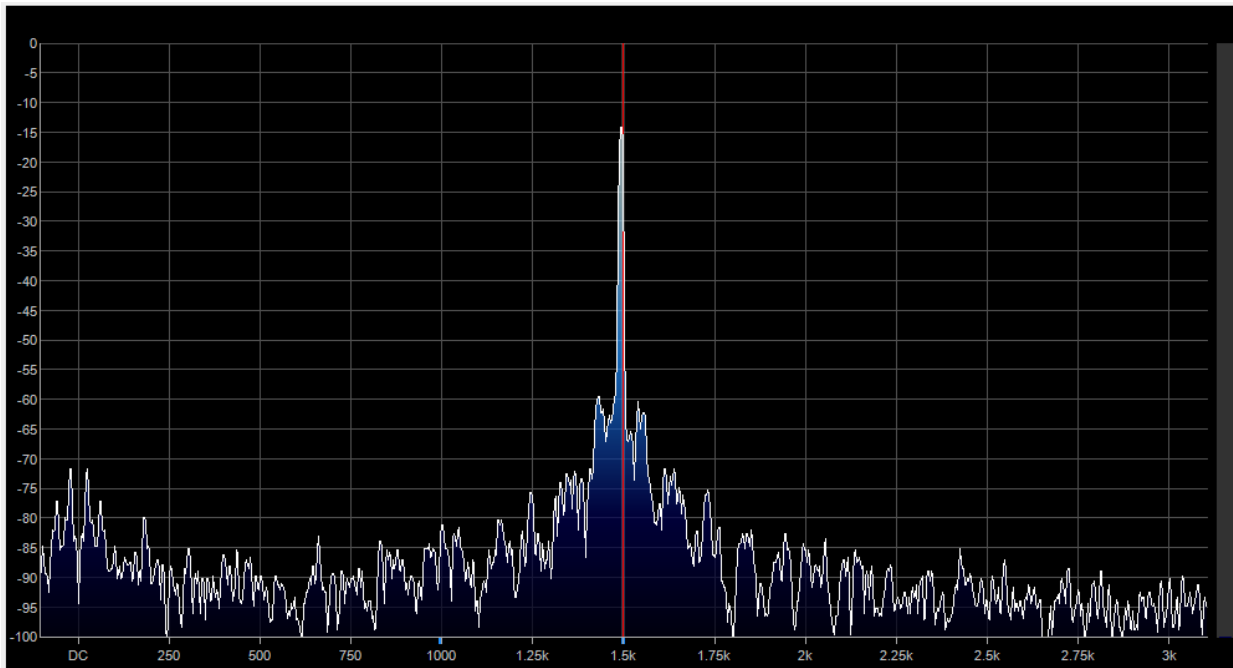NR = F1-1 Noise dropped 10db average (low side) and rolls off (high side) to -85 dbm
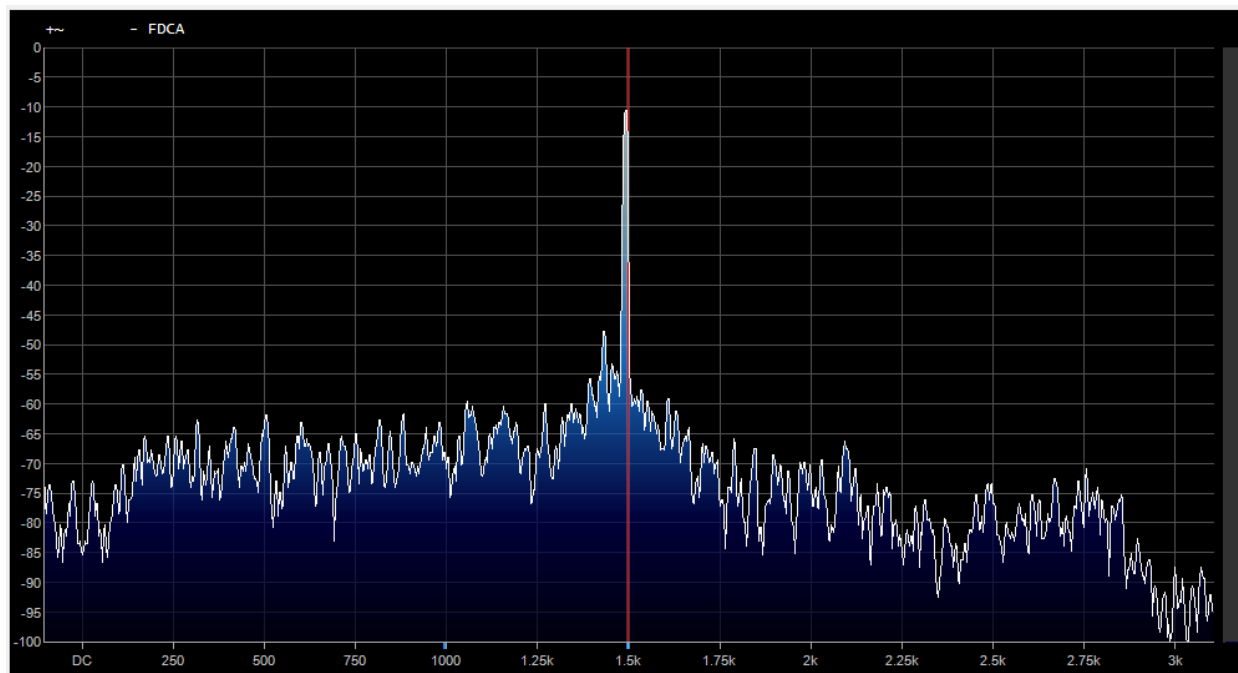
NR = F1-2 There are peaks and valleys in the noise floor and this anomaly is repeated in other setting.
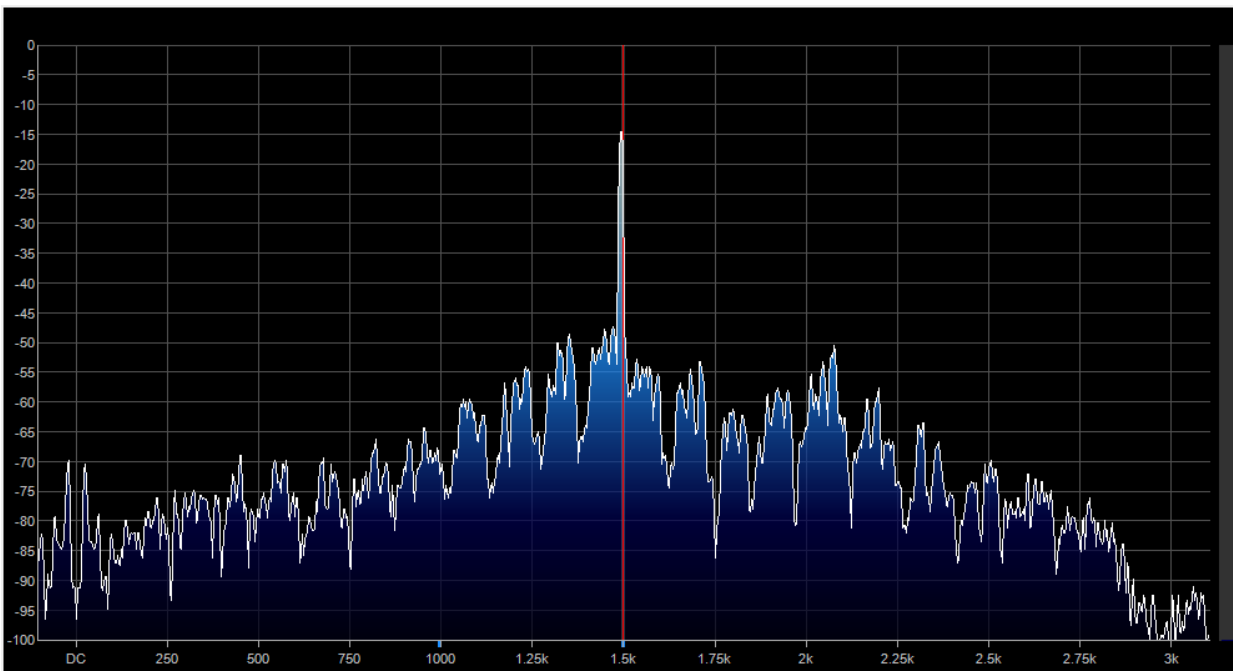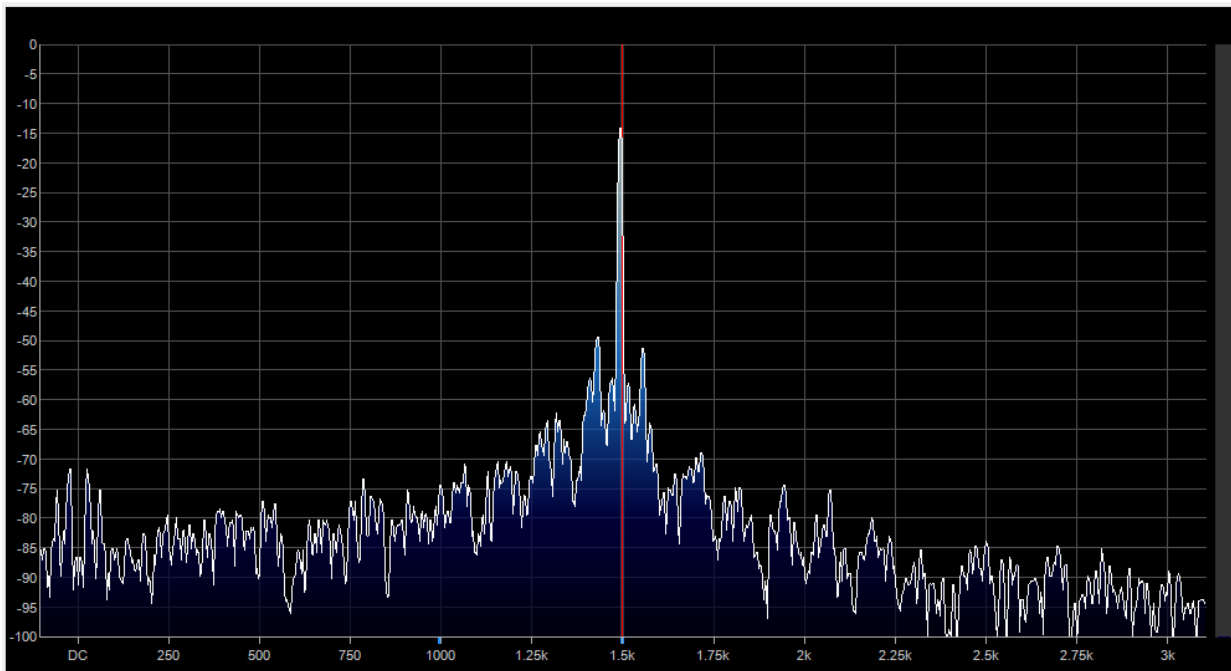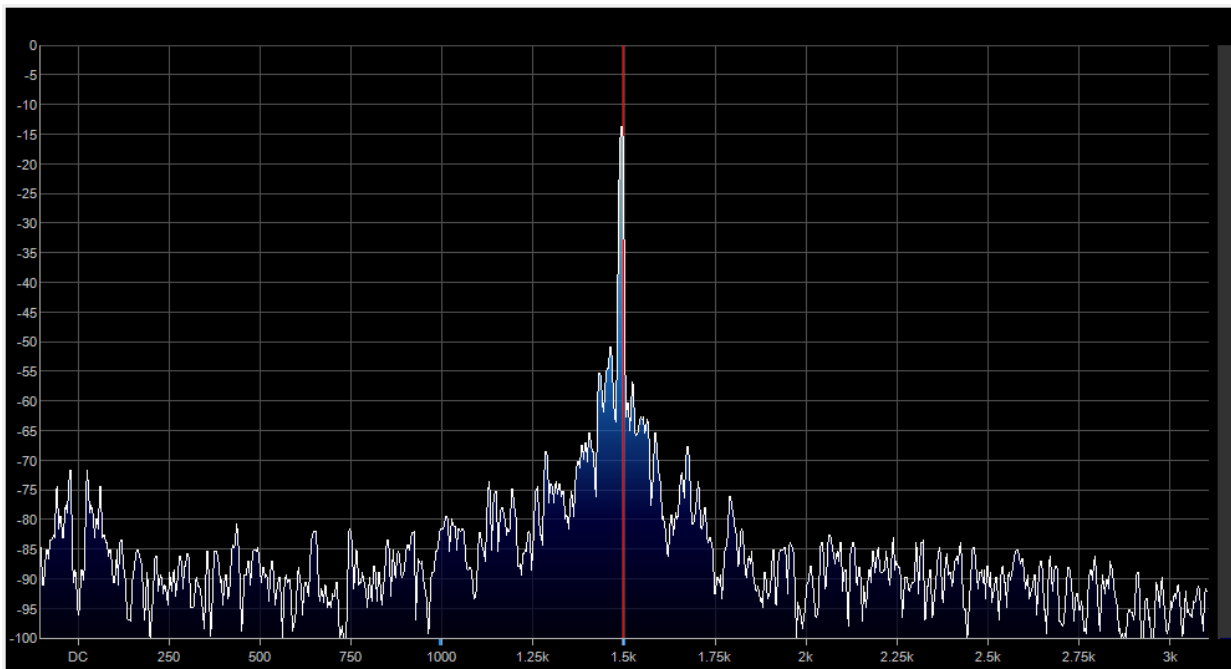


NR = F1-3

NR=F1-4 noise floor dropped considerably.



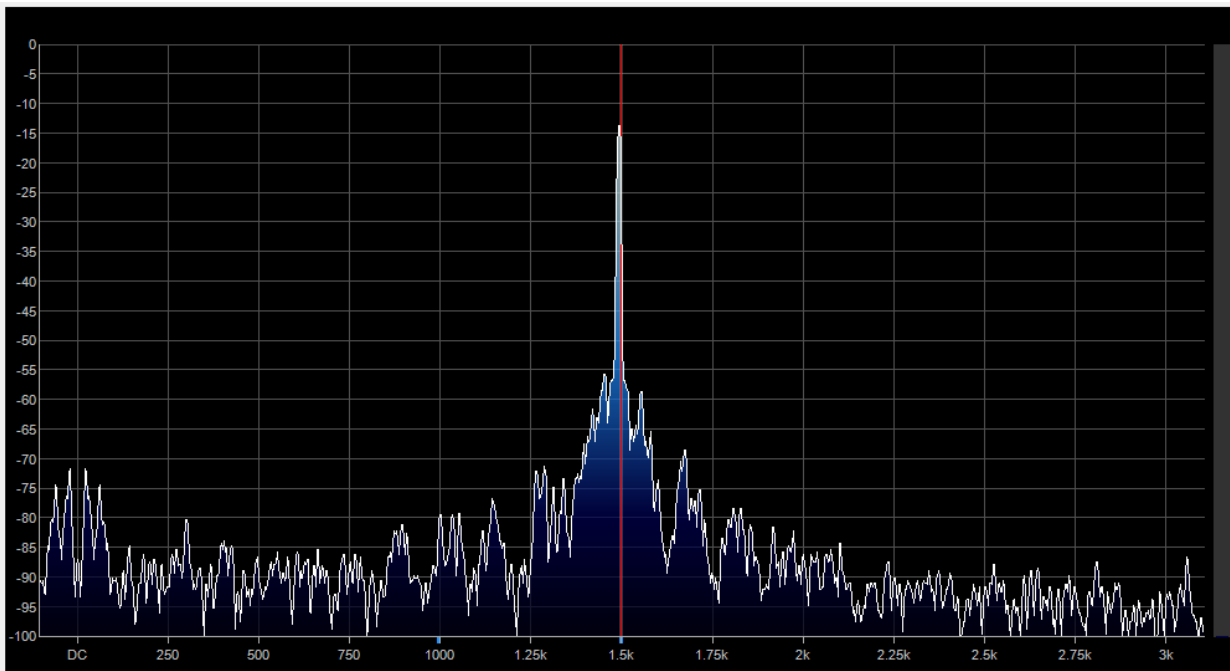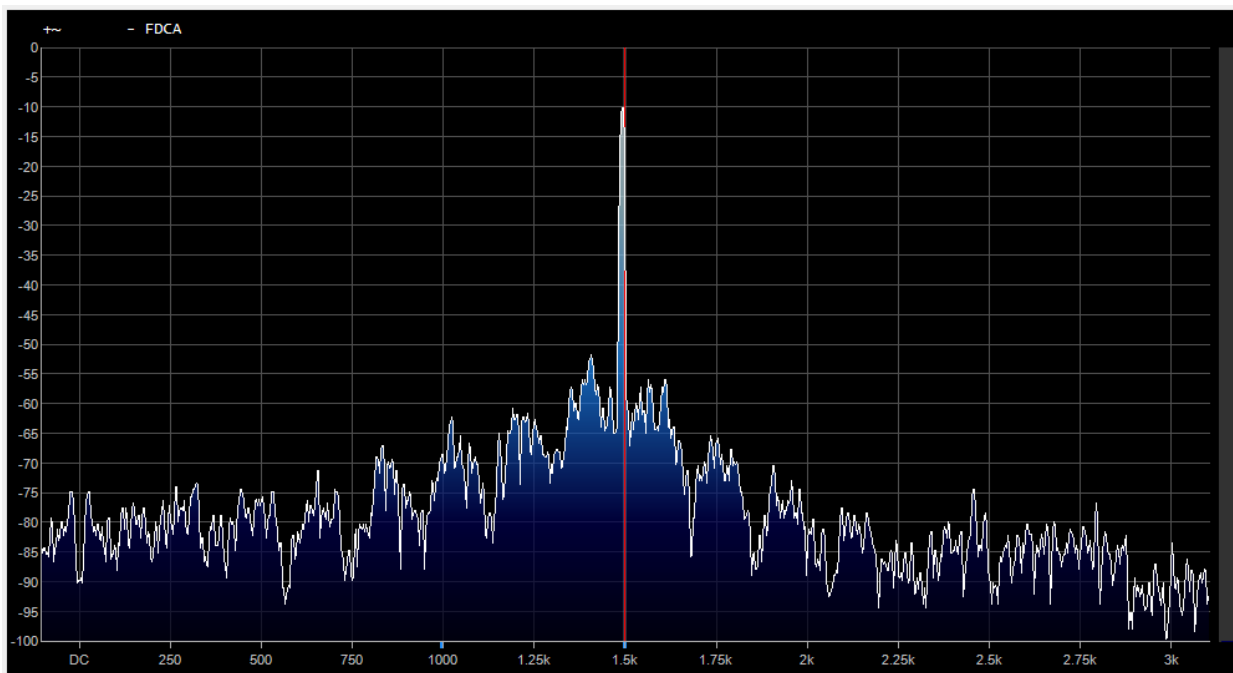NR = F2-1 Noticed that the peaks and valleys are spaced wider apart.
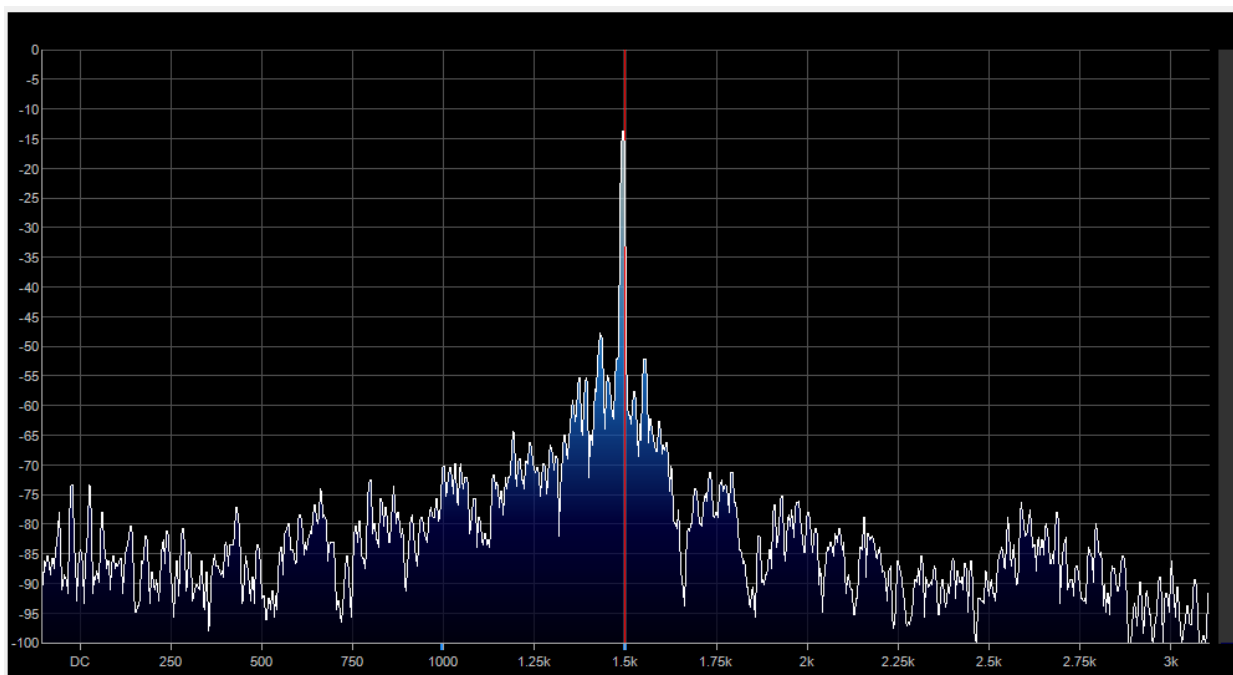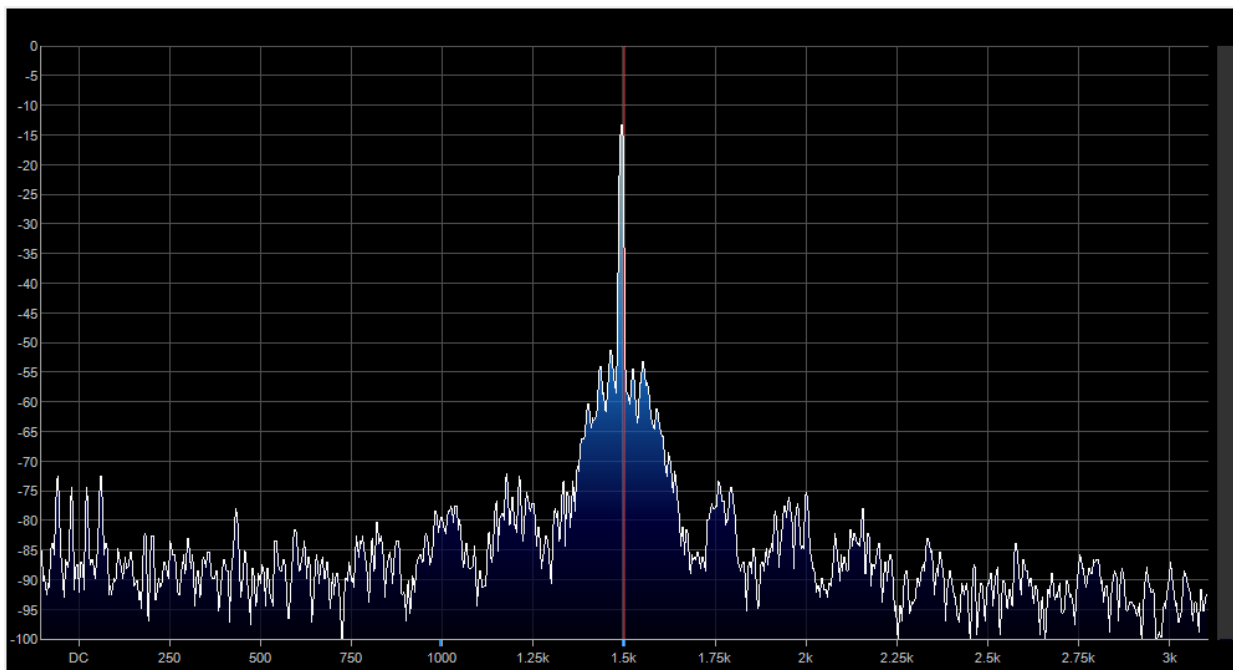
NR = F2-2



NR = F2-3  This setting and F2-4 are similar.

NR = F2-4



NR = F3-1.  The peaks and valleys are spaced even further apart.

NR = F3-2



NR = F3-3. This setting and F3-4 are similar.

NR = F3-4
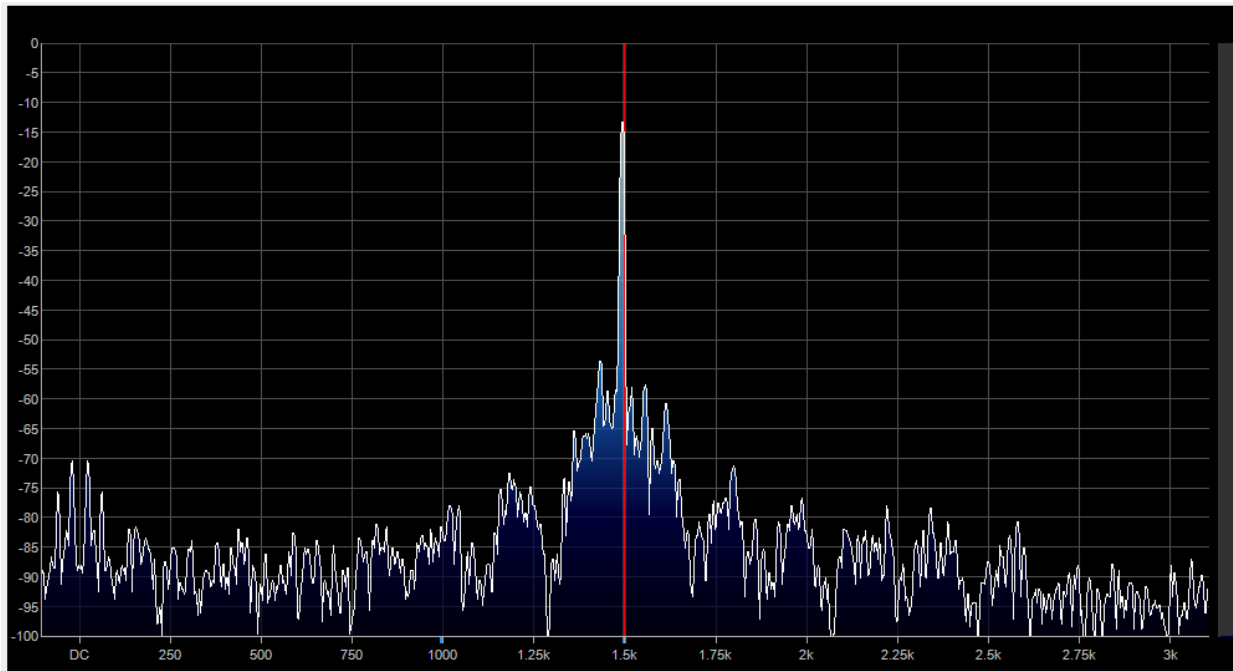


NR = F4-1 About every 250Hz, in the noise floor, the peaks and valleys are spaced the widest.
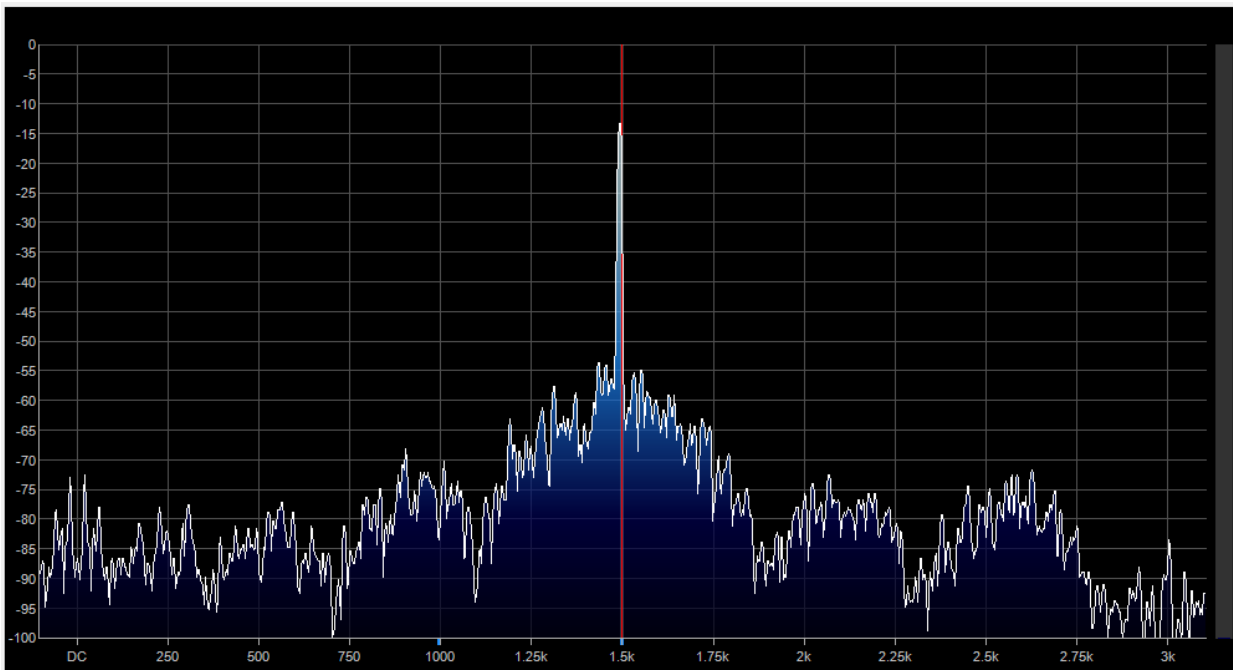
NR = F4-2
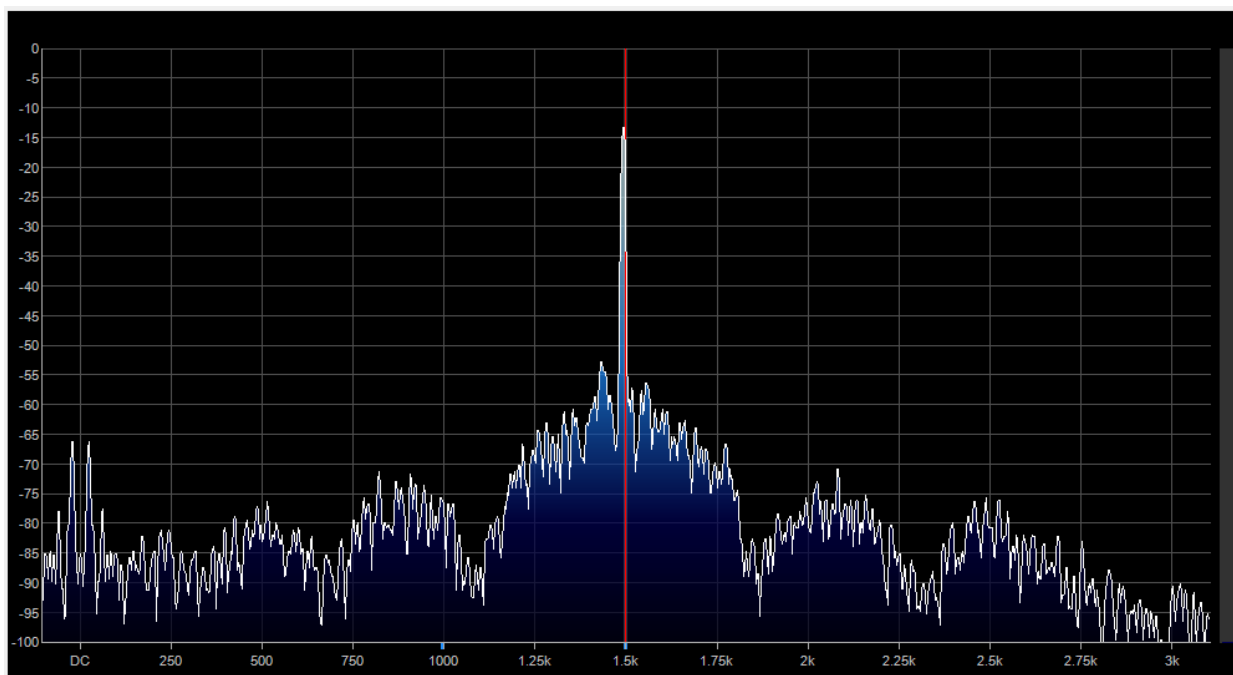


NR = F4-3
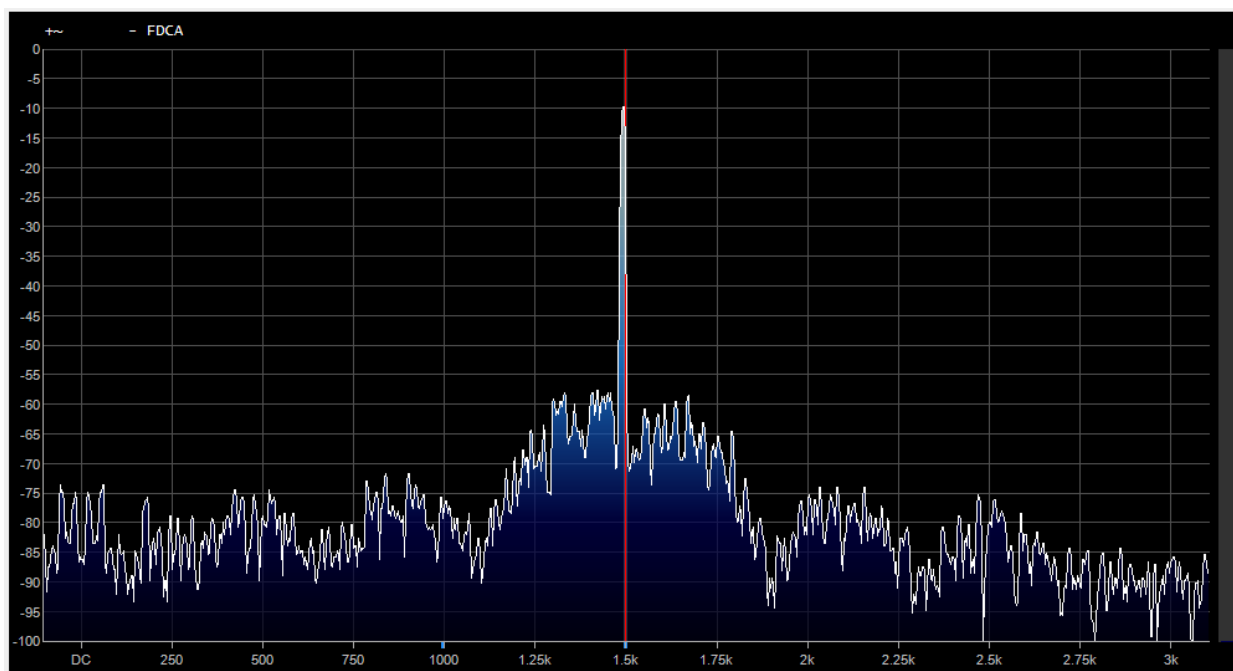
NR = F4-4
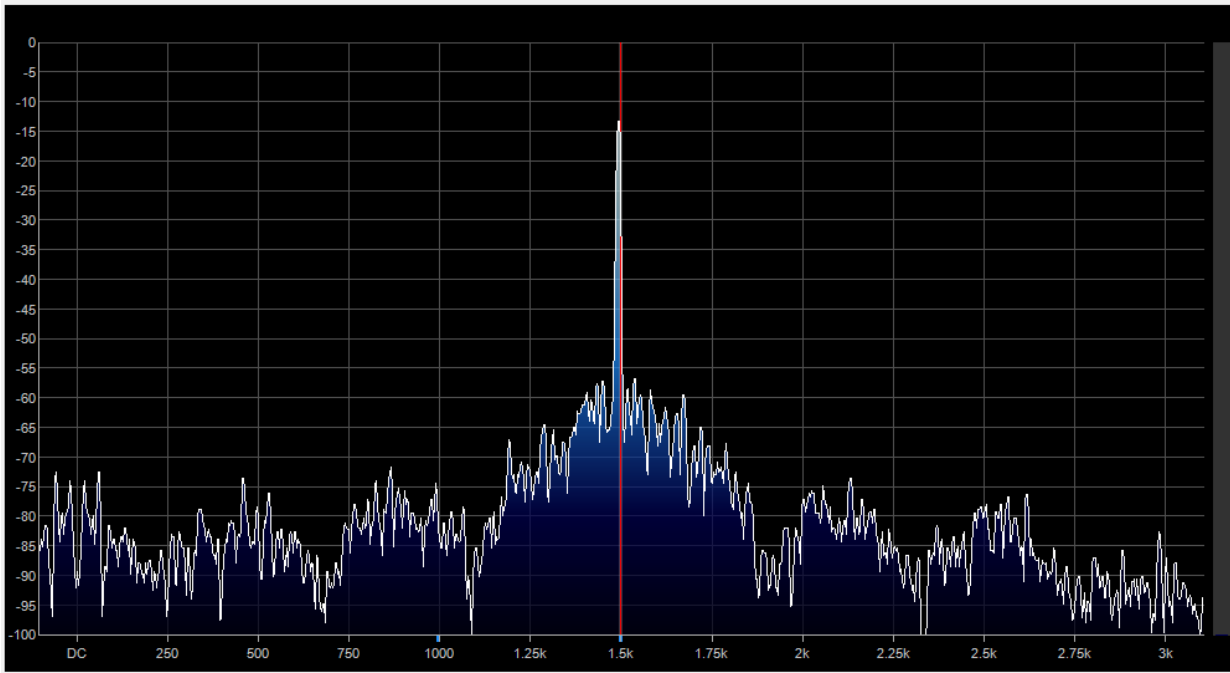


The next set of screen captures are the mix between the DSP-processed and unprocessed signals The NR settings F1-1 through F4-4 are mixed with the input.
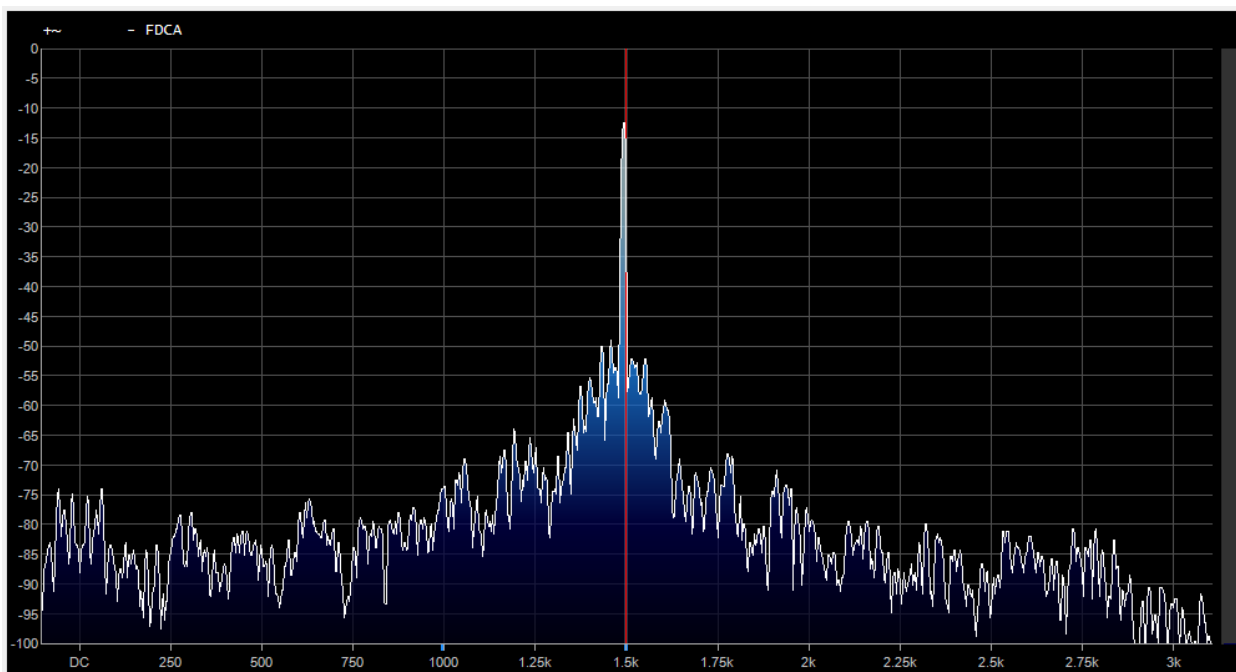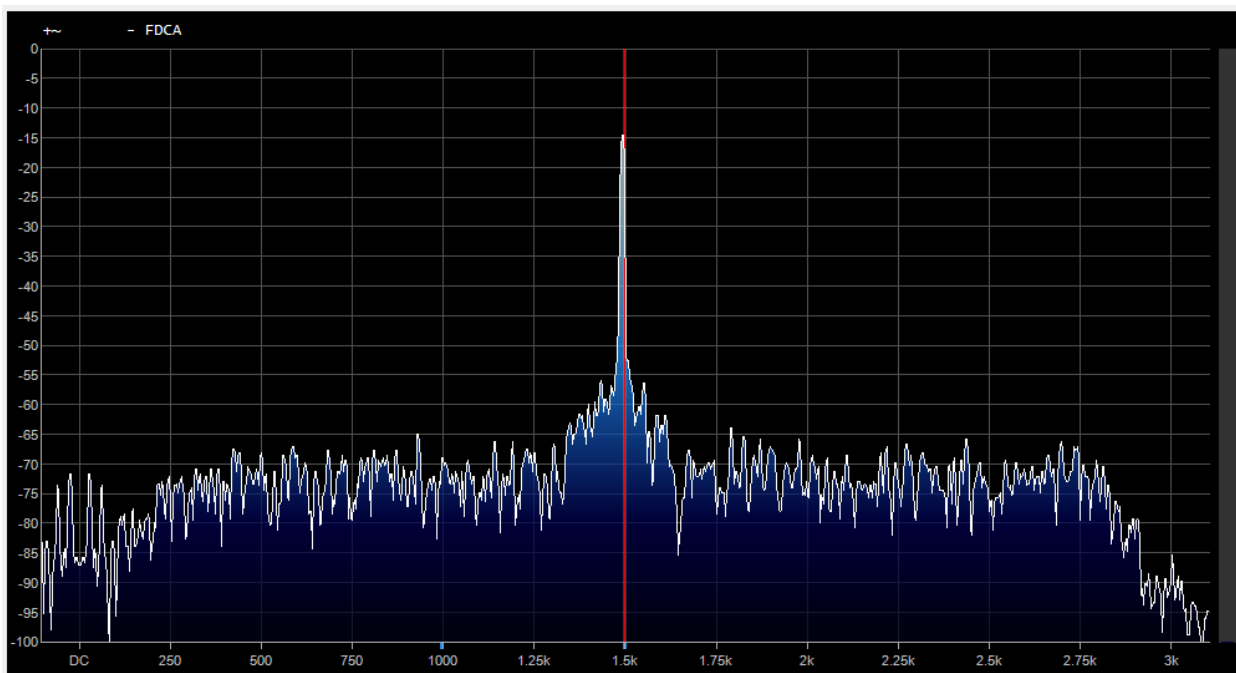
NR = F5-1
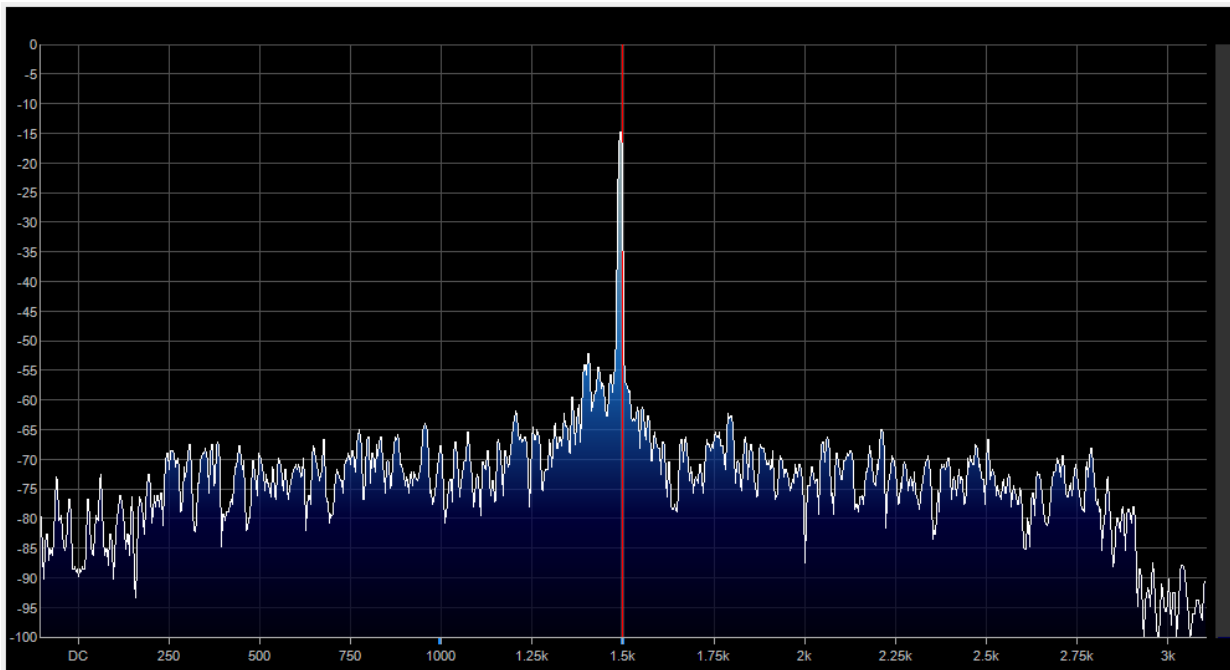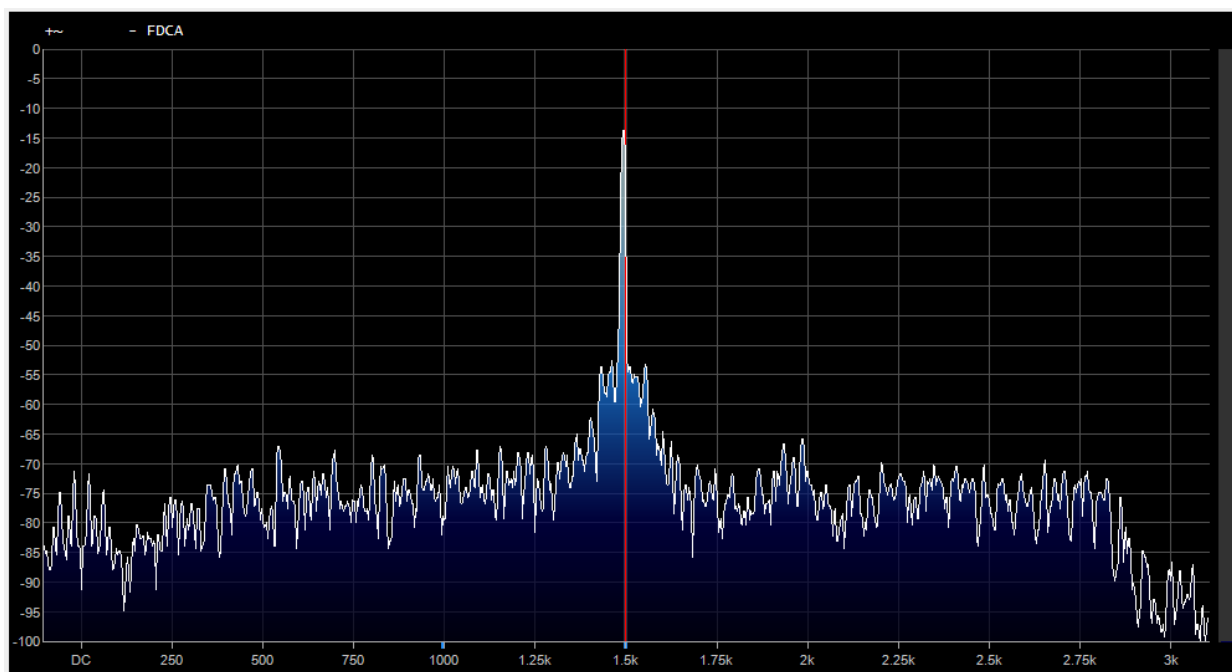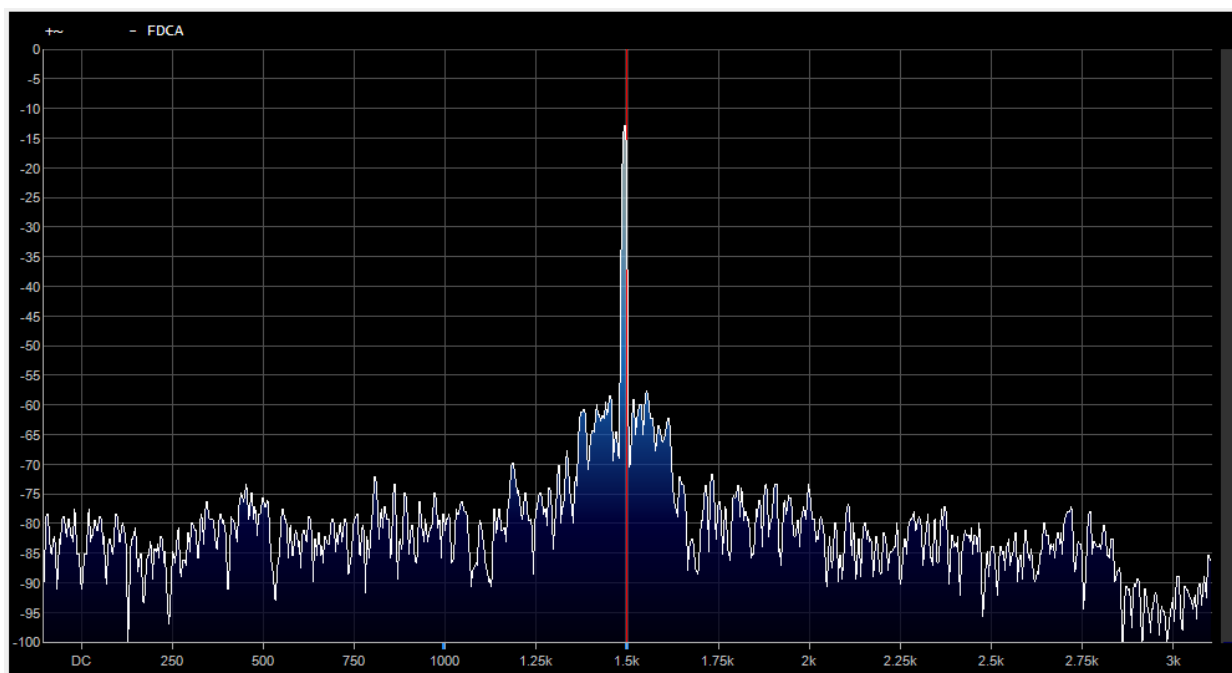
NR = F5-2
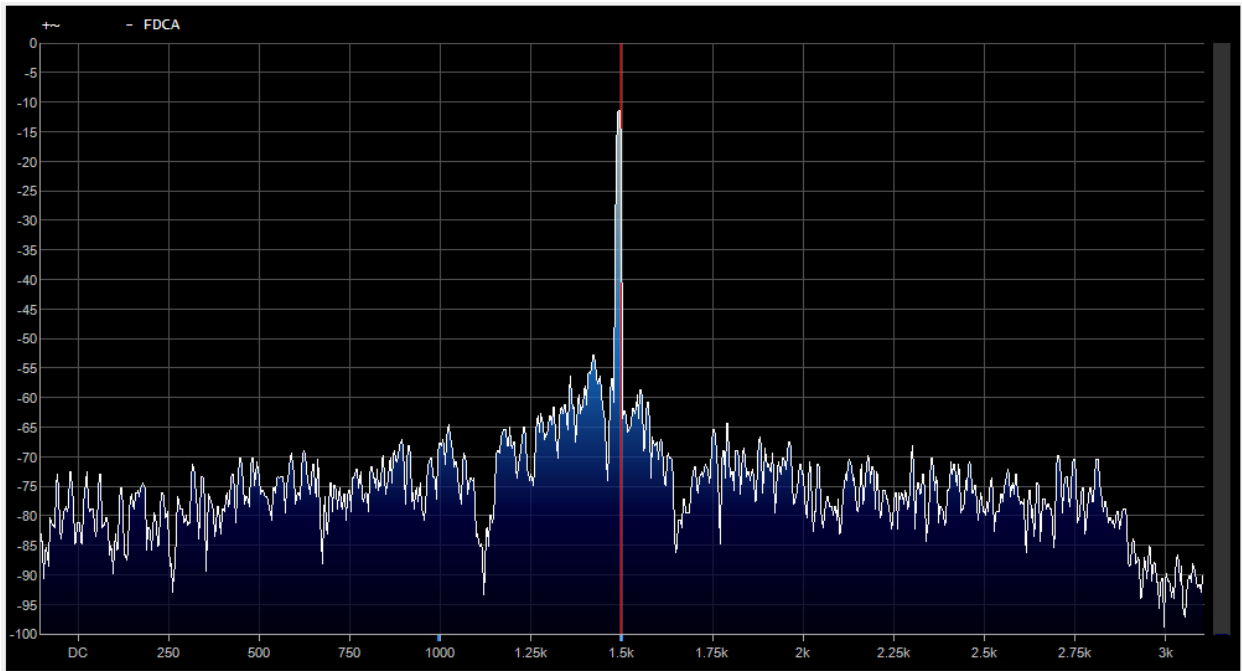


NR = F5-3
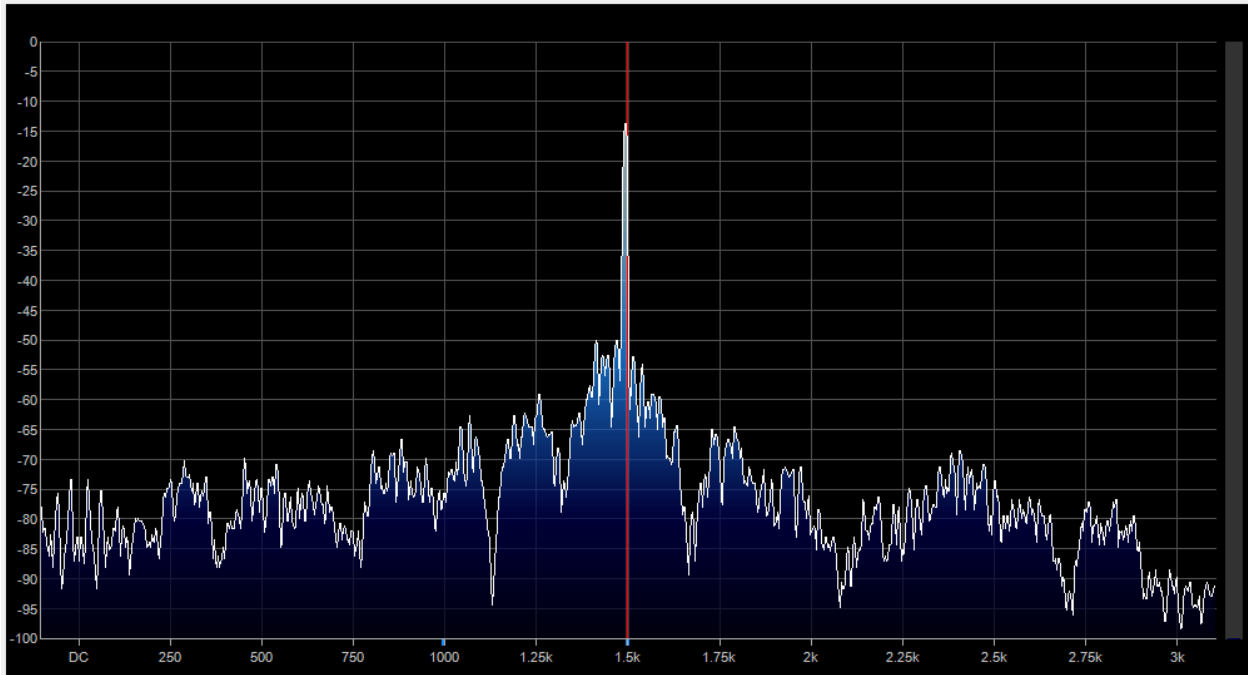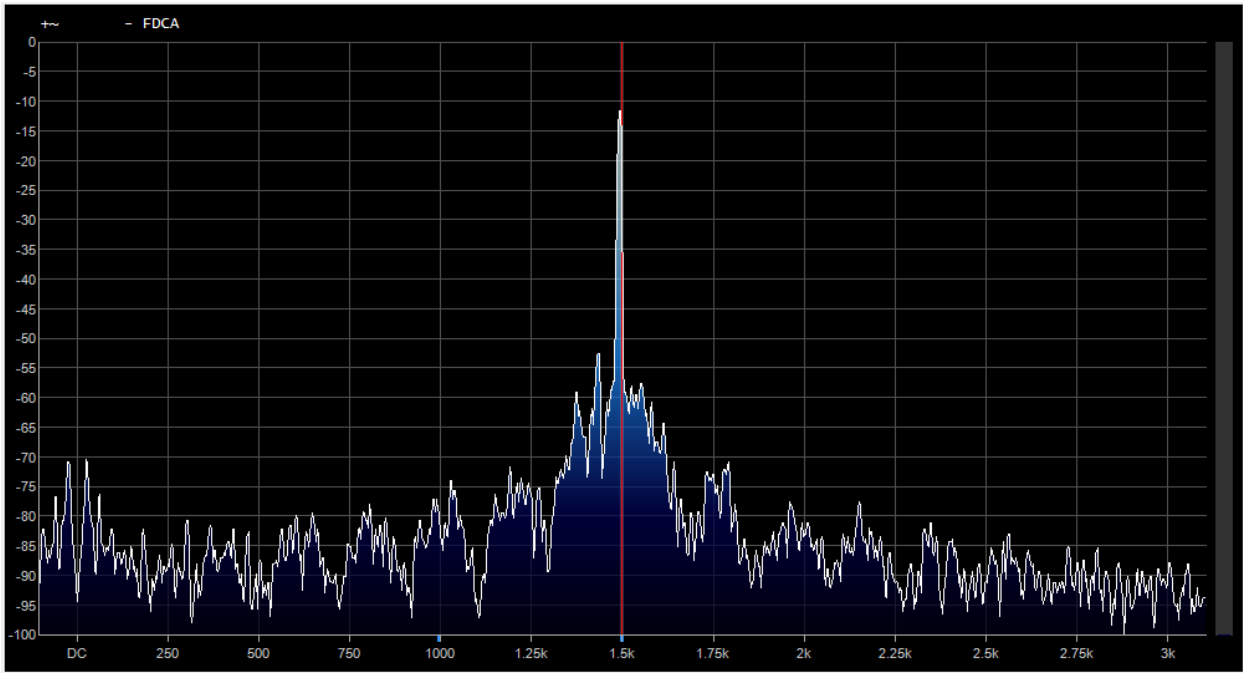
NR = F5-4
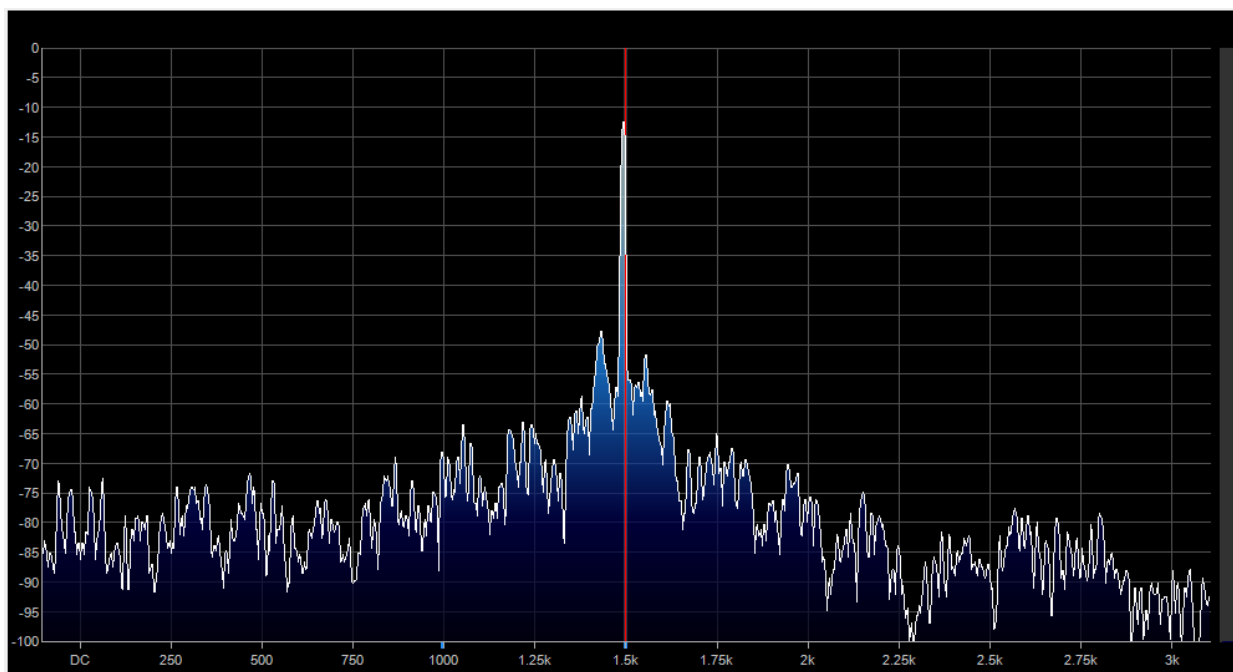


NR = F6-1
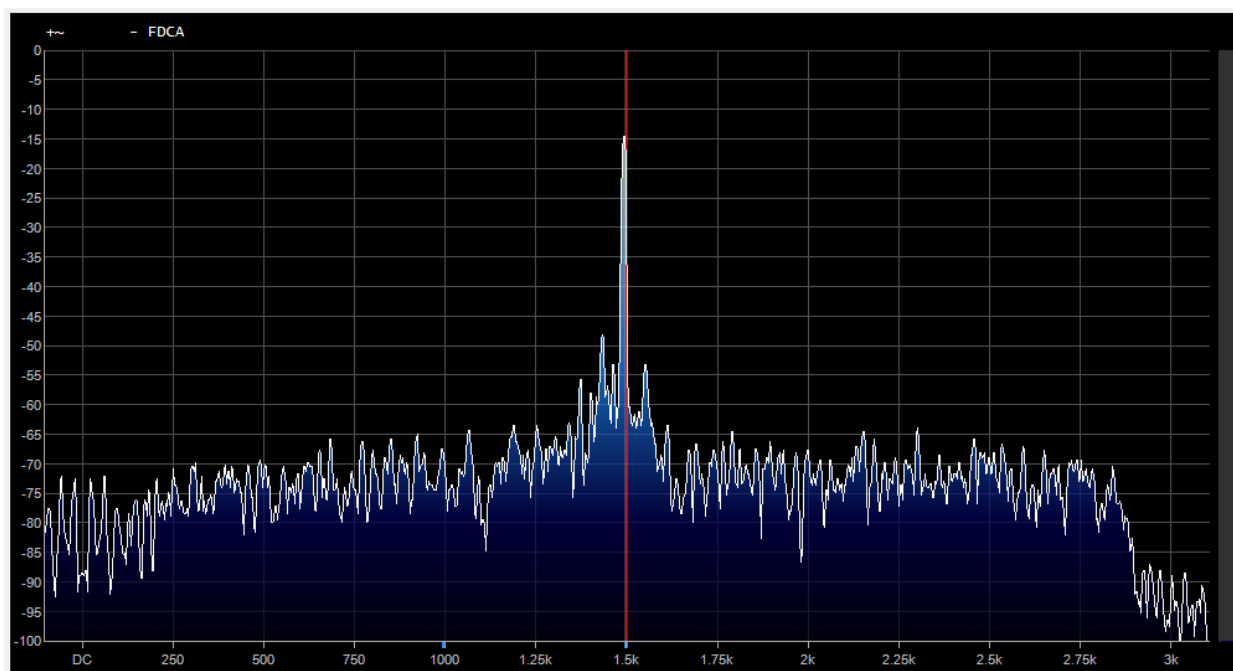
NR = F6-2
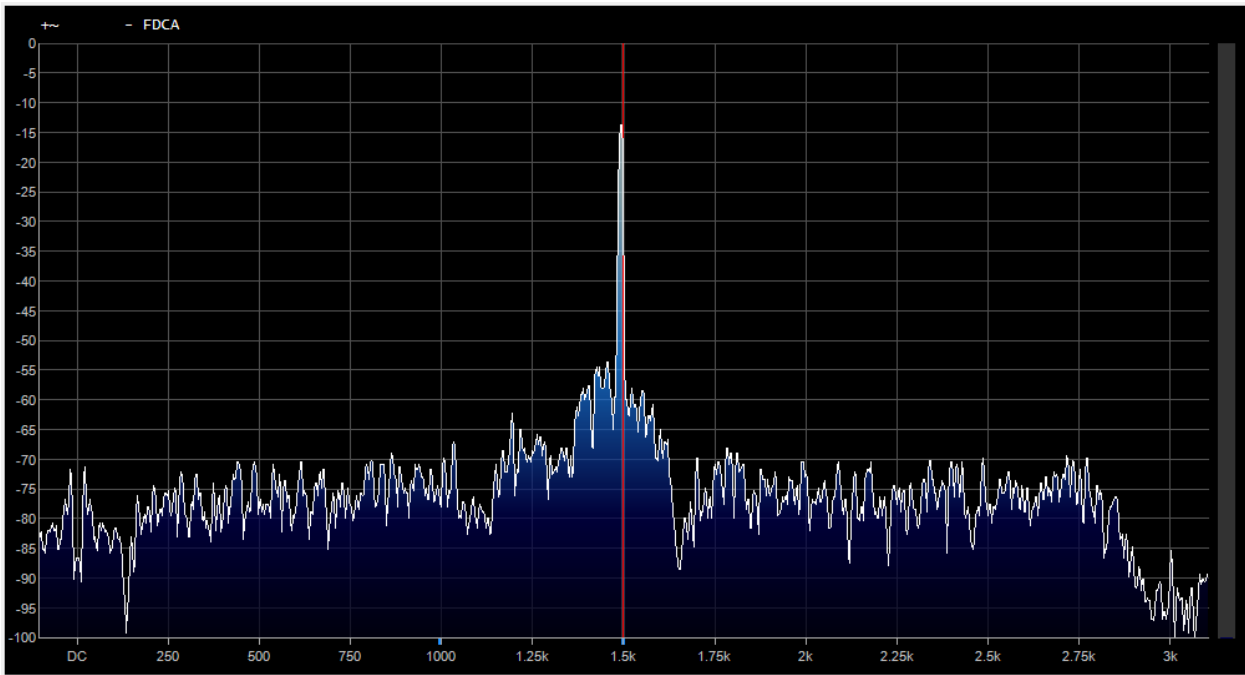


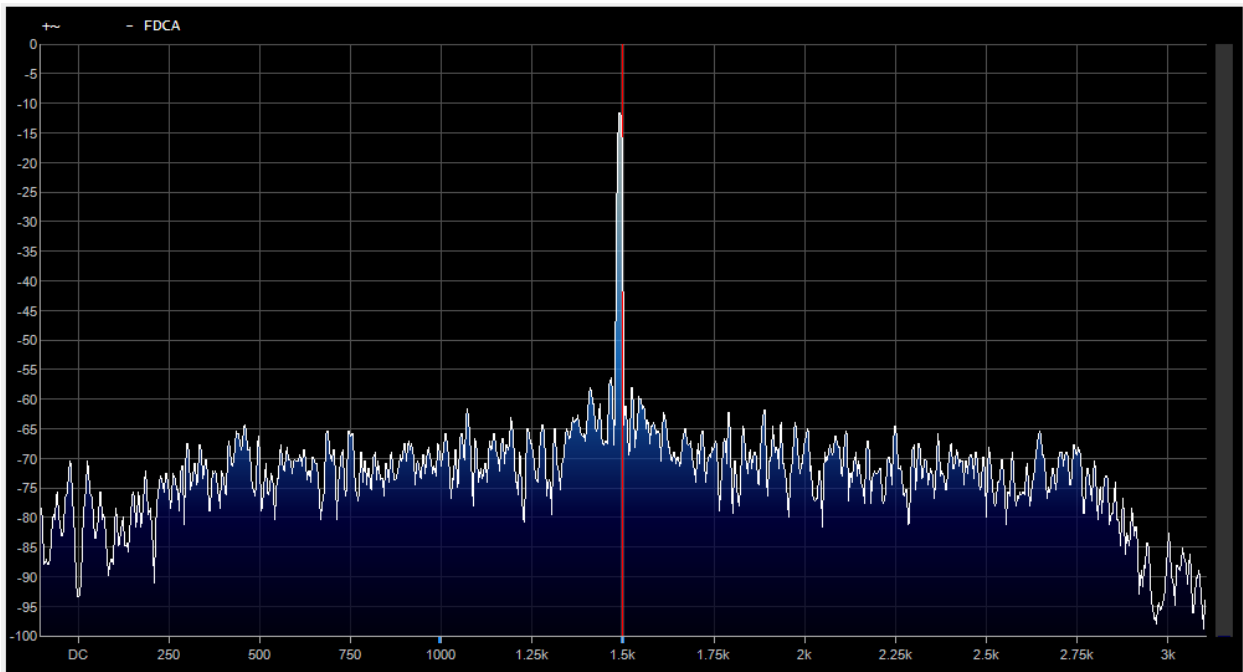NR = F6-3
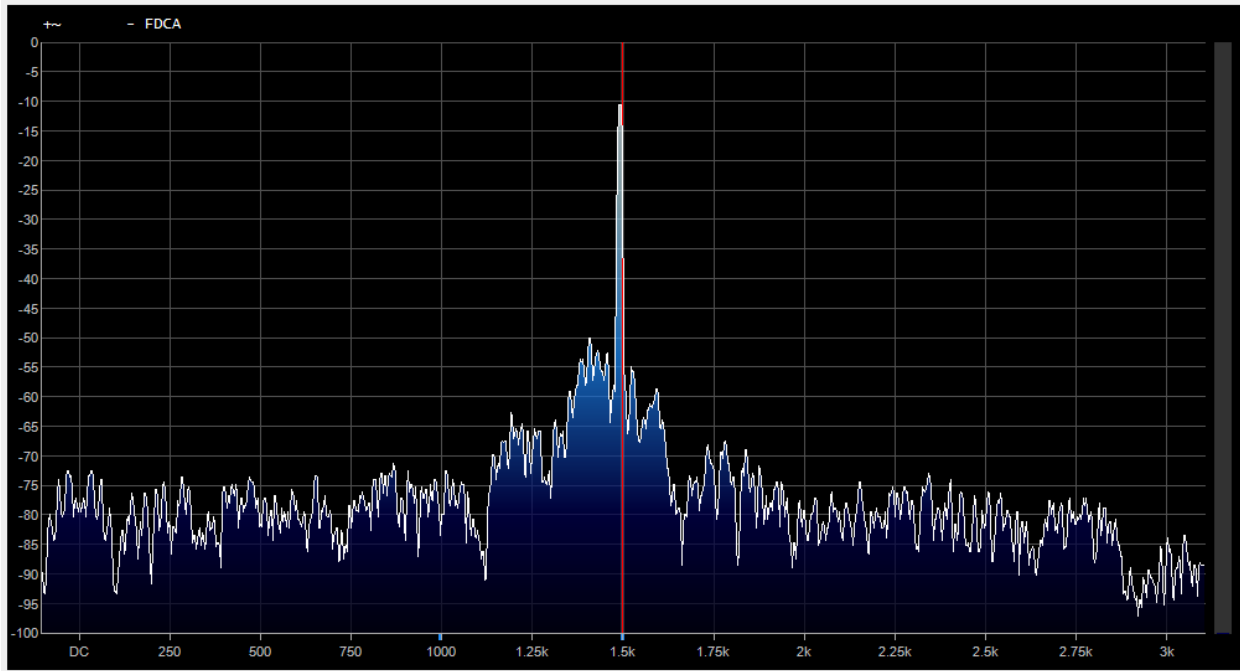
NR = F6-4



NR = F7-1

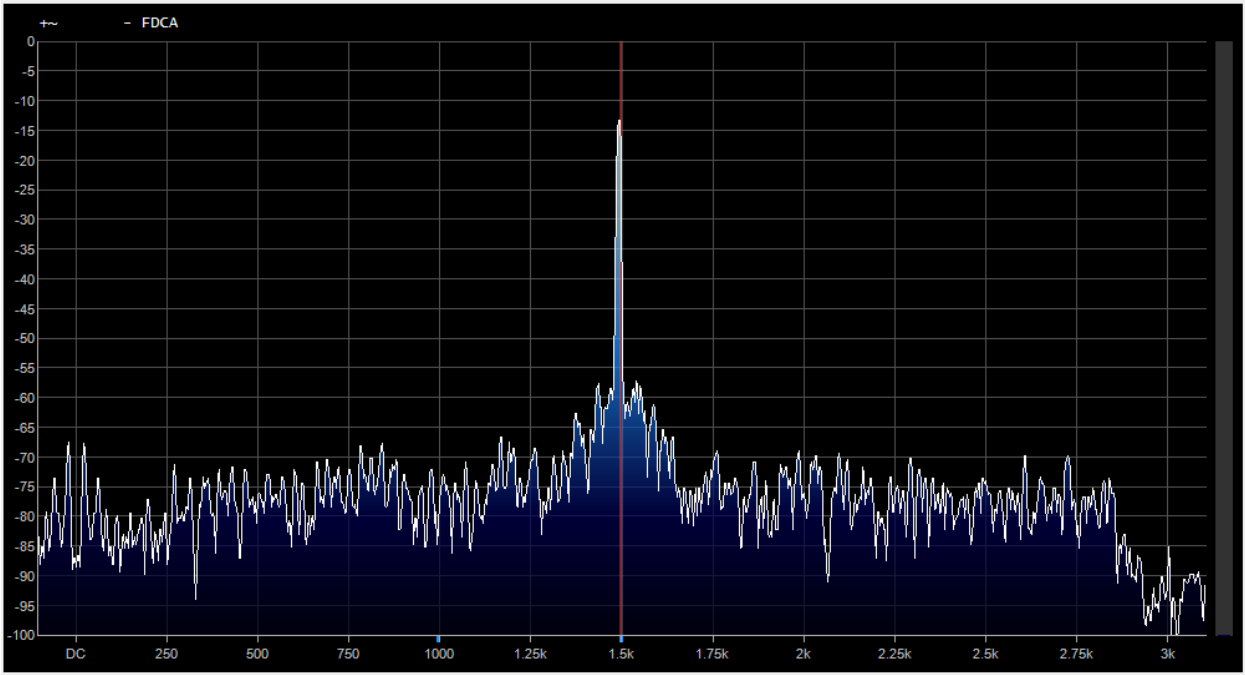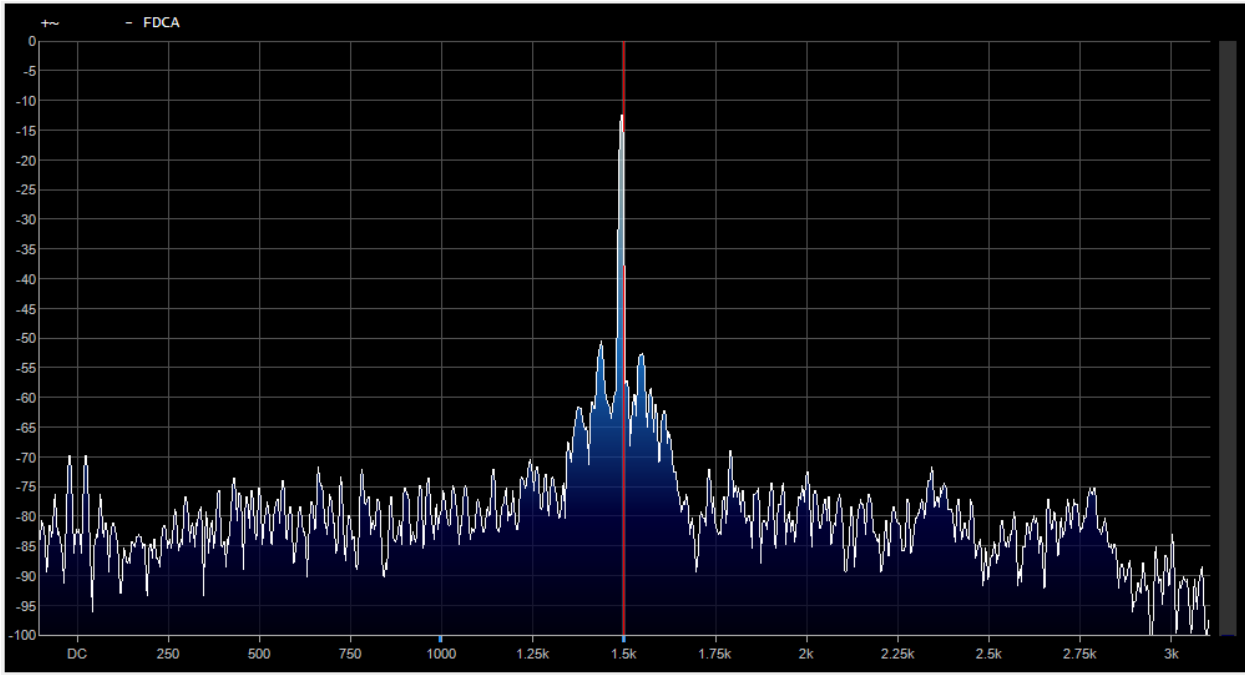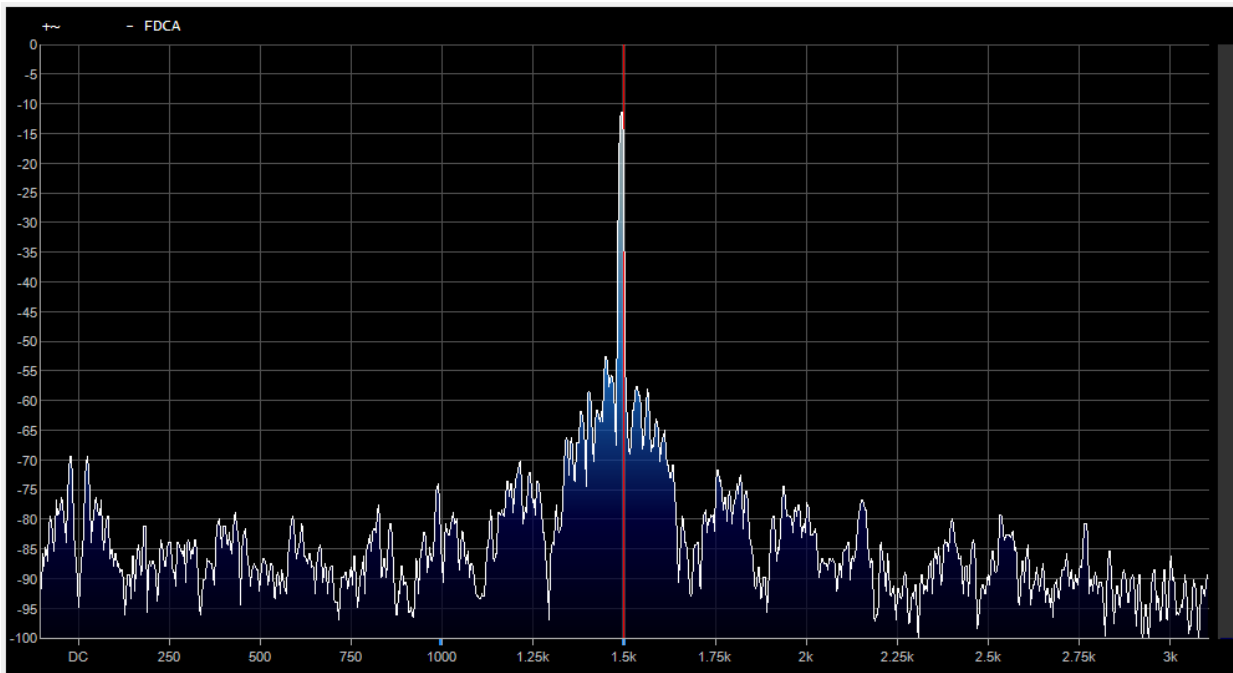NR = F7-2



NR = F7-3

NR = F7-4



NR = F8-1

NR = F8-2



NR = F8-3

NR = F8-4



CONCLUSION

Settings Fn-1 provides the least noise reduction.  Fn-2 and Fn-3 noise levels are almost the same and one setting could be eliminated.  Fn-4 provides the most noise reduction and having 3 settings, it would simplifying NR adjustments.  The NR delays that is, F1-n through F4-n, could be added to the configuration menu.  There are delay differences in the noise ripple but aurally, with a complex voice waveform, there's no distinction between the delay settings.

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