12th Convention on Digital & Radio Communications

Castelfeder 3-4 October 2010

SDT

Note the 'T'

A completely digital SSB exciter

Hardware by Giuliano I0CG, Software by Alberto I2PHD
English text by John F5VLF/G3PAI

What do we mean by SDR ?

SDR – Software Defined Radio or….

SDR – Software Defined Receiver ?

Often, unconsciously, we use the second definition…

But with SDR techniques, based on software, it is possible to produce a fully digital transmitter, where the input signal is first of all digitised and then used to produce an SSB transmission, with digital conversion to the final frequency.

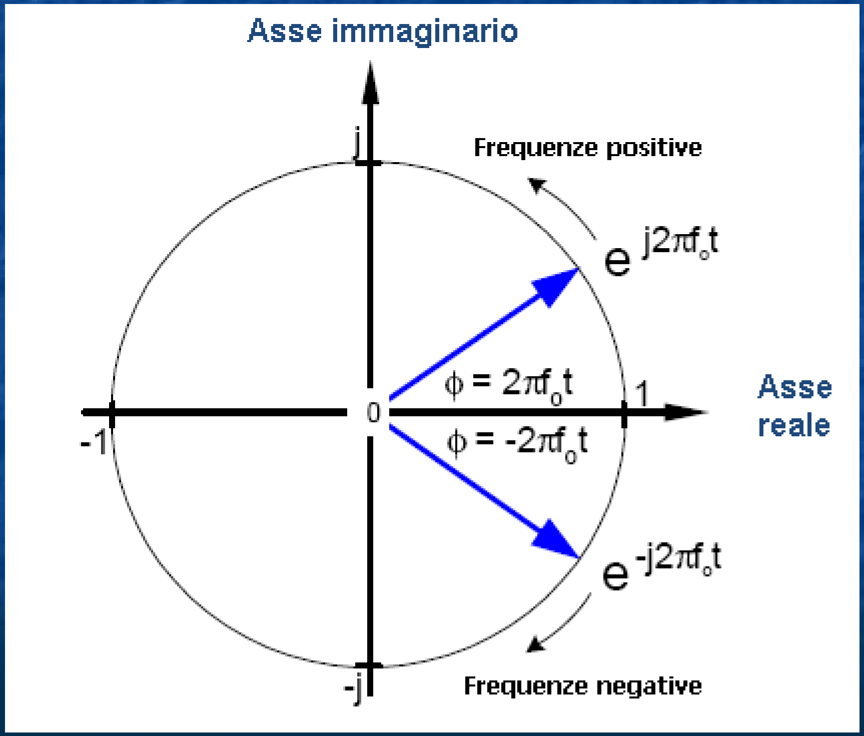
In practice

Digital
Up
Conversion

D U C k - Donald Duck….

SSB “modulation” is really a translation of the input signal from baseband to the final transmission frequency

But it is not enough to consider positive frequencies. In this case we must also include negative ones….



So, taking into account the negative frequencies as well, this is what is really going on:

Pity the LSB has appeared, as we don't want it…

How to get rid of it? Obviously by software…

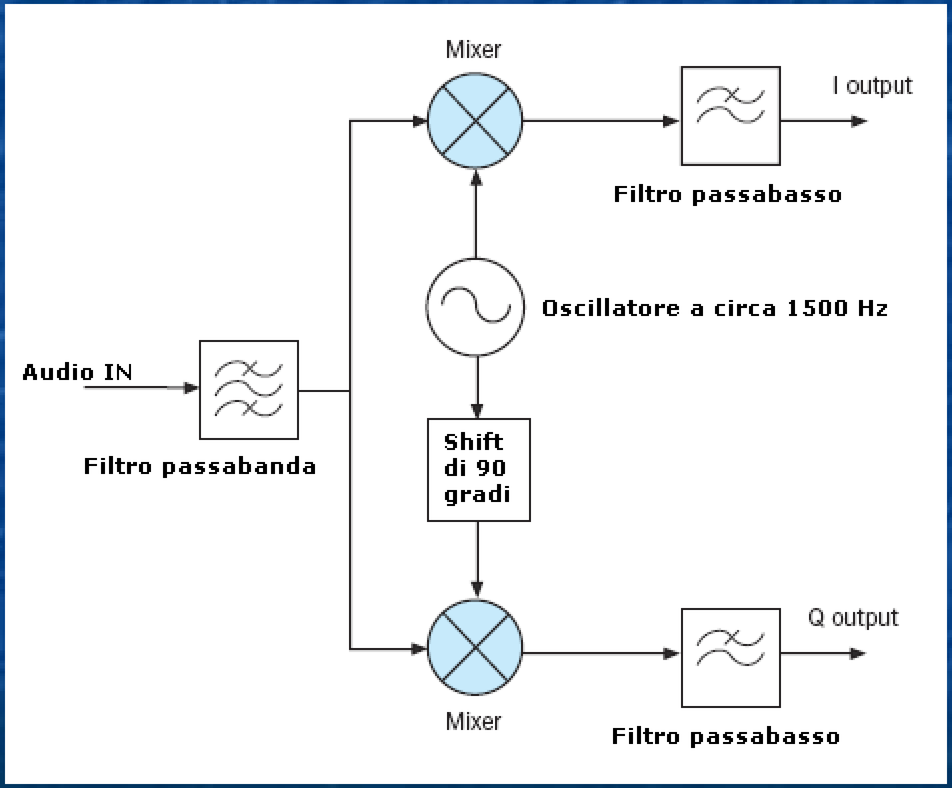
Quartz filter? You are joking! That's not software…

Phasing method ? Perhaps…. but this involves producing the LSB and then cancelling it out…

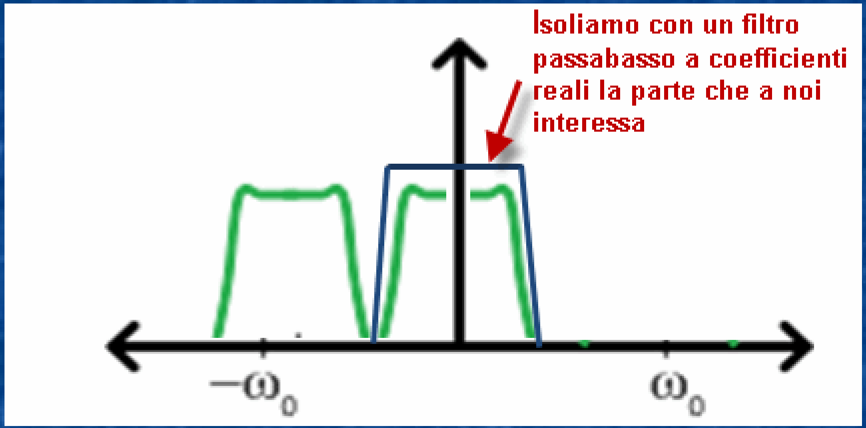
But wait a minute… what about the third method … what was it called ?

Weaver…

If we centre the USB on the zero of the frequency axis with a quadrature mixer so as not to have imaginary responses…

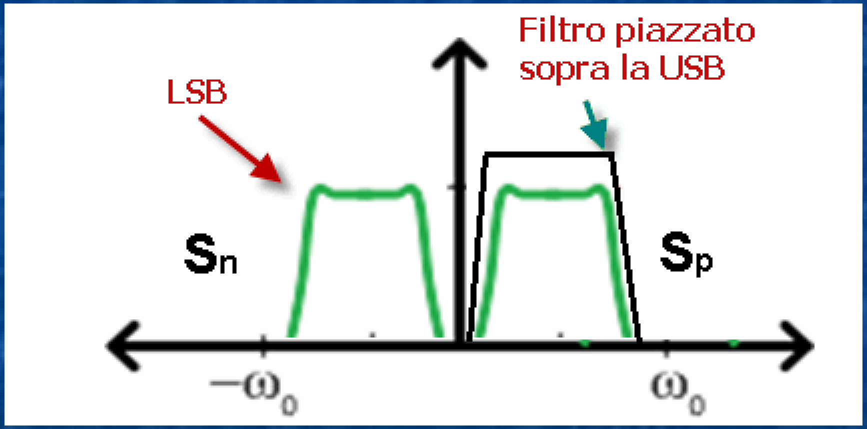


... and if we remove the LSB with a real lowpass filter (which when viewed in the complex plane is really bandpass)...



OK, but do we have to perform two multiplications for every input sample?

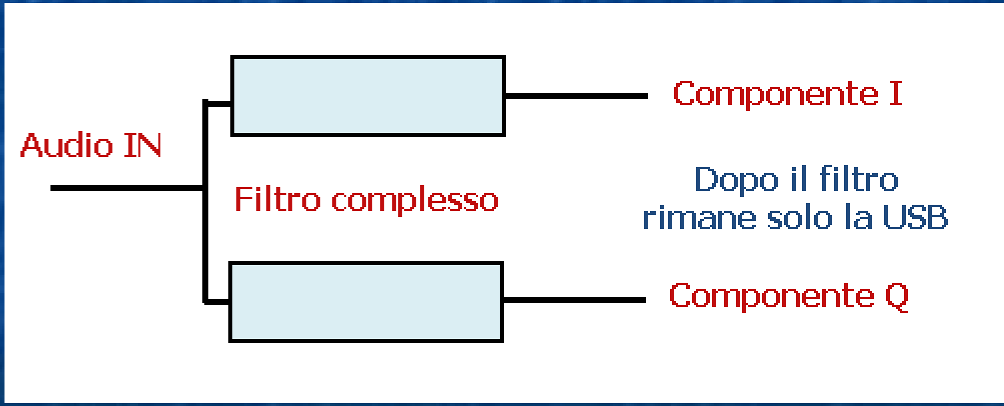
Suppose, instead of switching the signal, we switch the filter?



I prefer this idea … if we save a little

(so long as we *do* save) …

Clearly the filter now has to have complex coefficients …

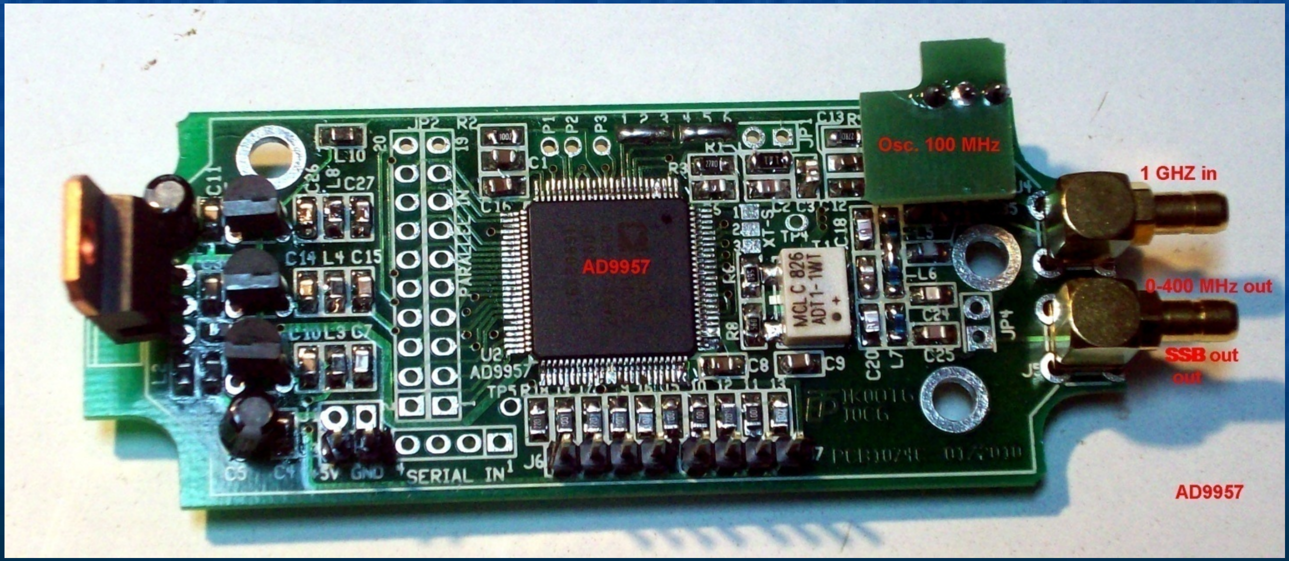


Fine. Now we have the two components I and Q which describe the analytical signal representing the USB, but they are at baseband … they have to be shifted to the final output frequency.

If for example we wish to transmit on a frequency of 100 MHz, a modulator is needed with a half-complex mixer with a sampling frequency of at least 250 MHz…

Fortunately there is a similar requirement in the world of commercial communications, and for every such need there is usually a company to provide a solution ….

PCB produced by Giuliano I0CG



AD9957 block diagram

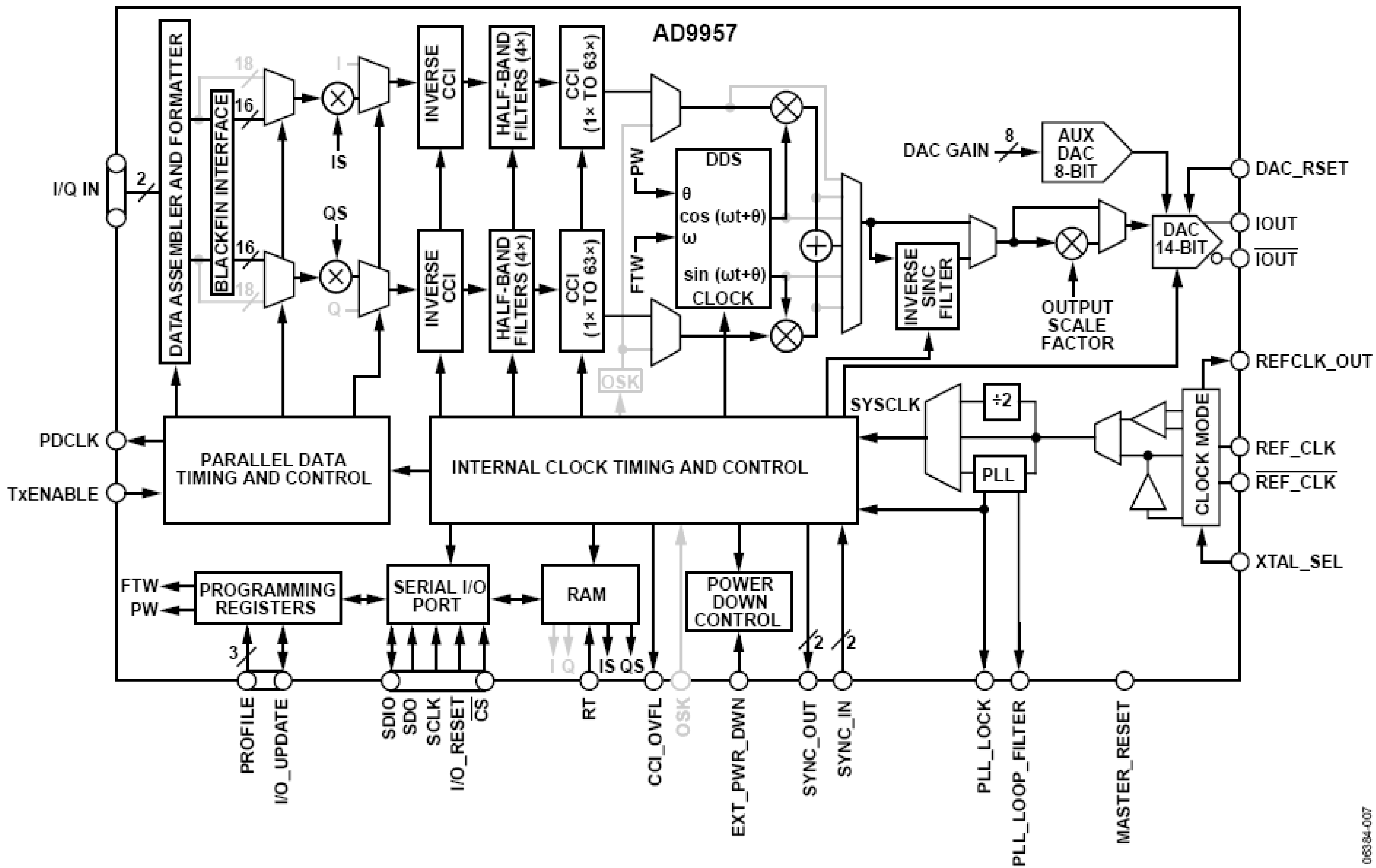


Figure 27. Quadrature Modulation Mode, Blackfin Interface

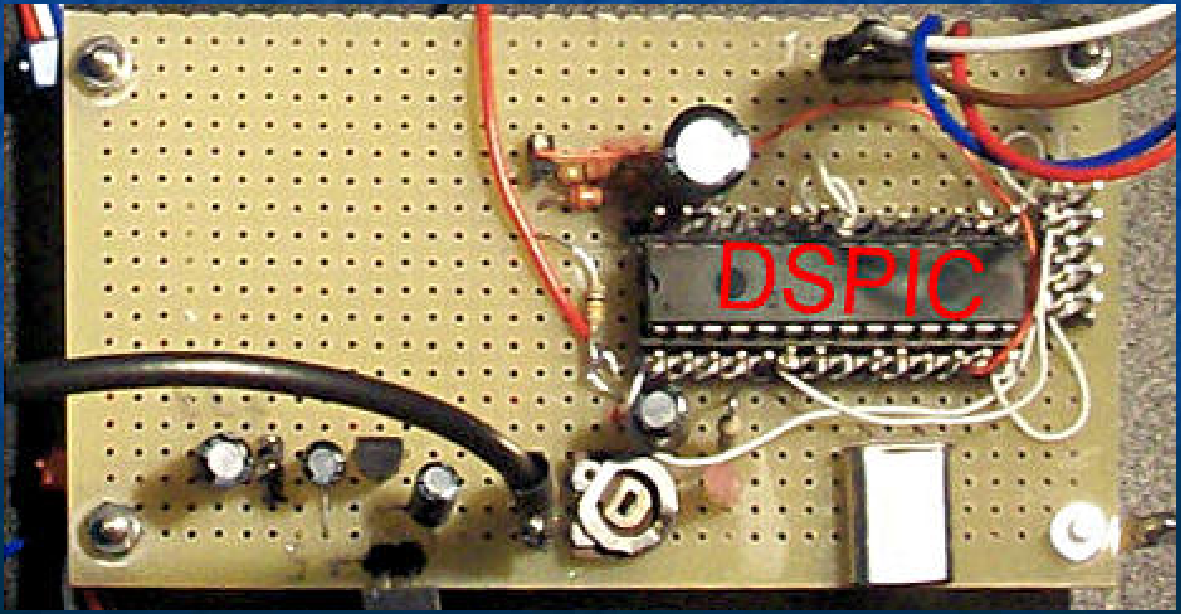
The AD9957 has a series of internal programmable interpolators, for ratios of up to 1008. The input signal should therefore be in I/Q format with a sampling frequency of :

2.5E8 / 1008 = circa 248016 Hz

The best way to do this is to sample the audio at 1/16th of this rate - that is circa 15501 Hz - filter at that frequency with the complex filter previously mentioned, and then interpolate x 16

We can do it with a convenient low-cost component, the dsPIC from Microchip. A fairly conventional PIC, with an on board ADC and a DSP core capable of 50 milion MAC operations per second. Not bad for a 5 dollar chip…

And you can also get it in a 28 pin DIP version…



Advantages of the dsPIC

- Compatibility with a standard PIC
- Free compiler (demo version)
- Components cost only a few euros
- Available in 28 pin DIP
- The high level of integration of this component, containing all the necessary peripherals, has greatly simplified the hardware on the circuit board which now is essentially just the dsPIC.

Advantages of the dsPIC (2)

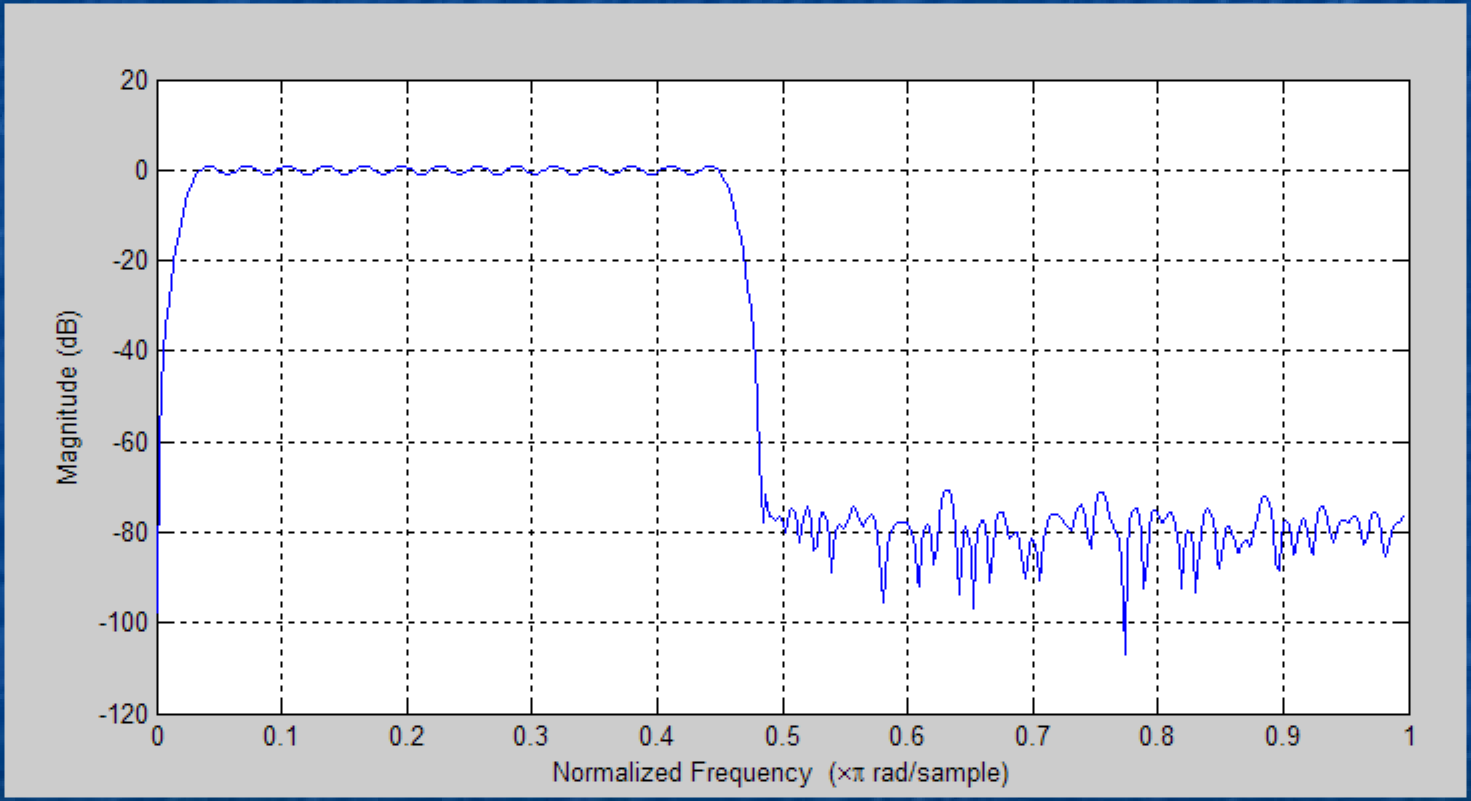
Inside the DSPIC33F128GP802 used for the project

- 128kb flash memory: 2% used
- 16 Kword RAM: 80% used
- 12 bit ADC used for sampling the microphone signal
- 16 bit ADC used in the debug phase (to observe the I/Q signals in analogue mode)
- 2 SPI interfaces used to transfer I/Q digital data t the DUC AD9957
- 2 Timers to control the timing of sampling and transmission to the DUC

So, to sum up, the dsPIC digitises the audio at 15501 Hz, eliminates the LSB with a complex filter, and then perform a x 16 interpolatione, bringing the sampling frequency to 248016 Hz.

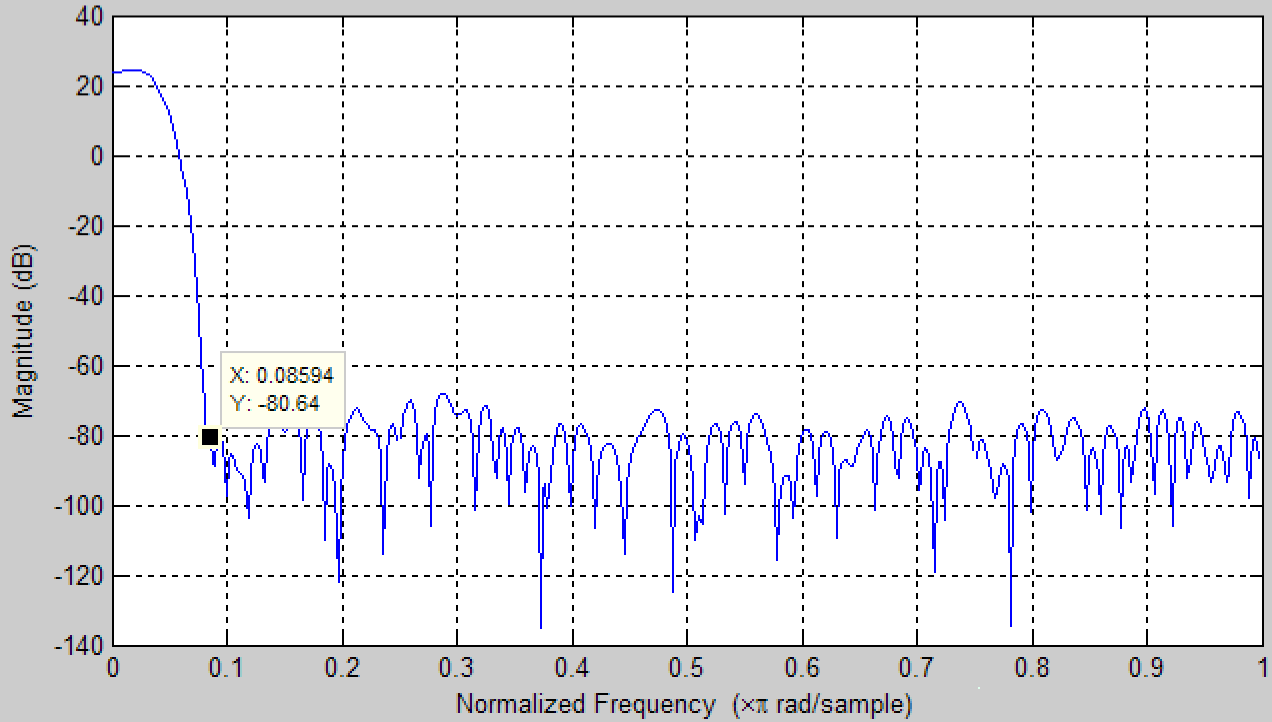
At this point the I and Q samples are sent, in Q15 format, to the AD9957 through a double serial interface of type SPI (Serial Peripheral Interface), and the AD9957 does its dirty work. It produces USB (or optionally LSB, changing the sign of the imaginary part of the Numerically Controlled Oscillator), with an output frequency between, say, zero and 100 MHz, so as not to get too close to the Nyquist limit (125 MHz).

A few graphs showing the results obtained when designing the filters with Matlab



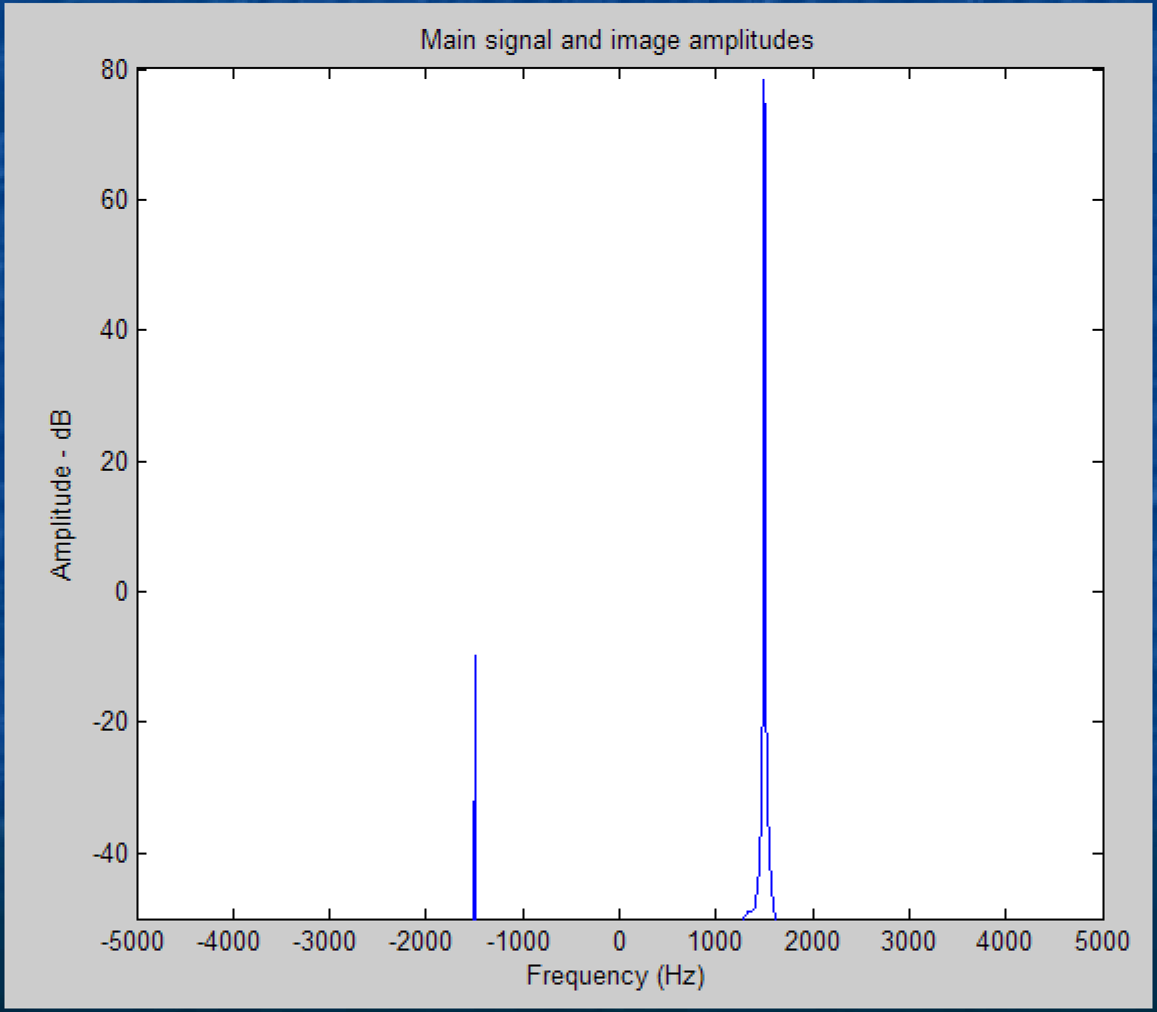
Variation with frequency of the output of the complex filter, taking account of rounding the coefficients to 16 bit

Response of the anti-aliasing filter needed for the interpolazione stage. The passband magnitude is at +24 dB to allow for the amplitude equalisation required for a x 16 up-sampling

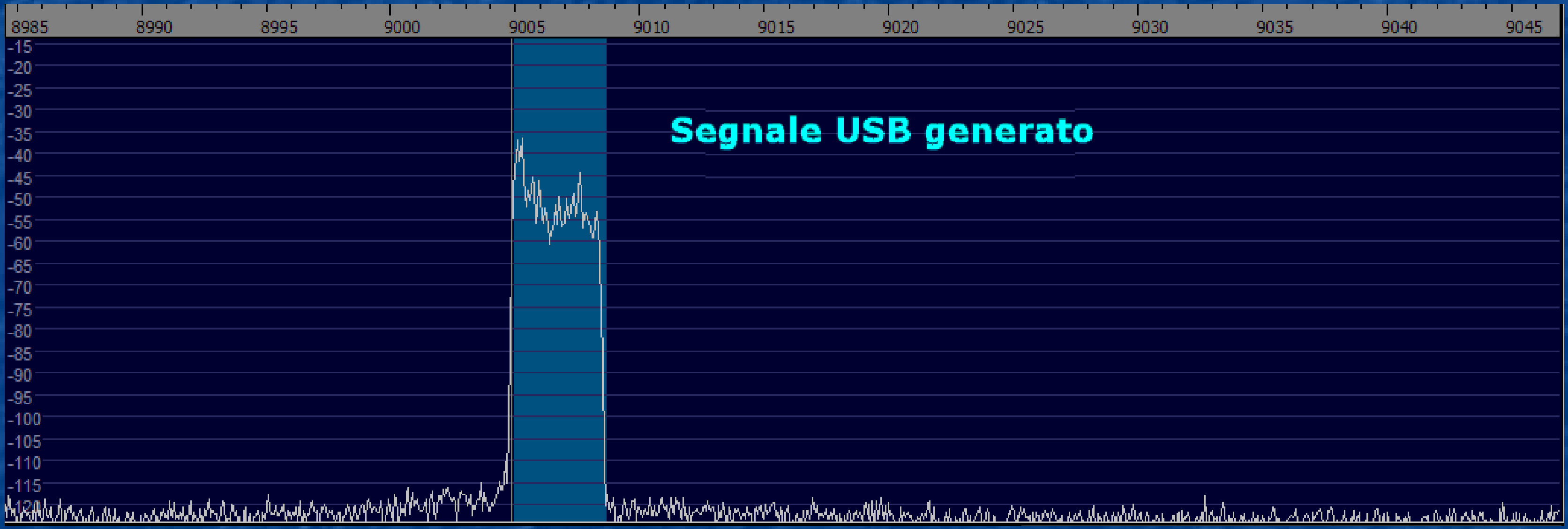


0.08594 * 248016 / 2 = circa 10657 Hz, ben al disotto del valore richiesto per questo filtro anti-alias.

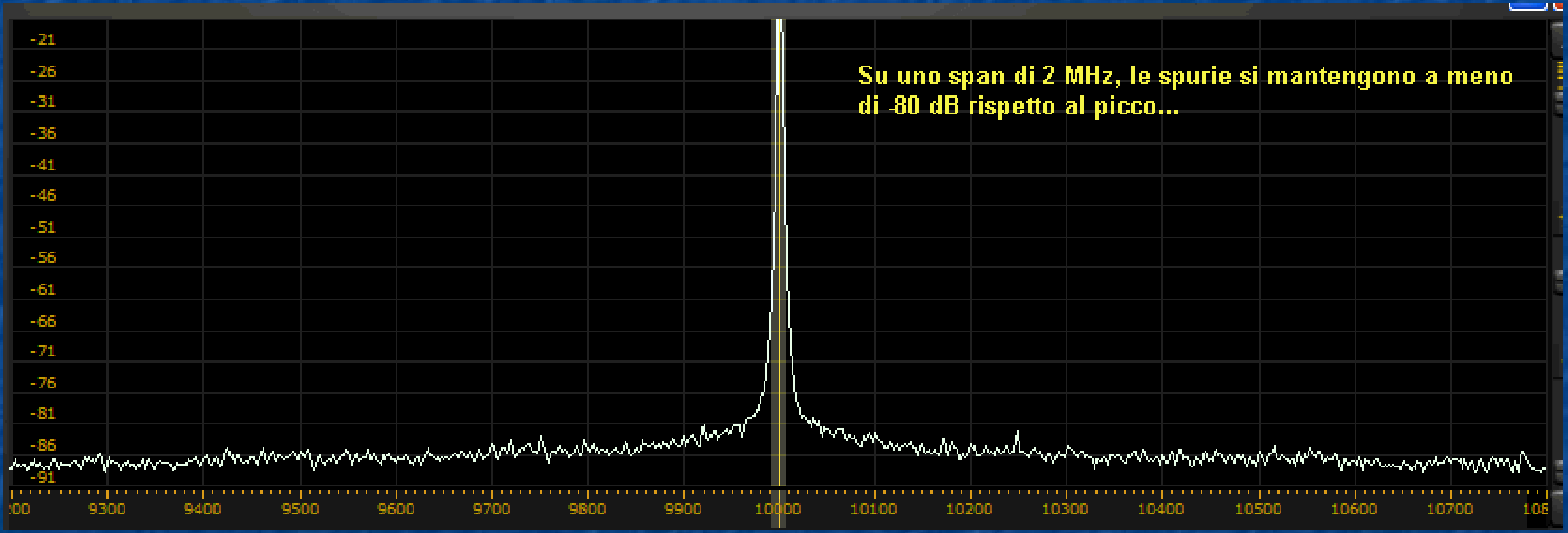
Simulation with Matlab of the rejection of the unwanted sideband. With 1500 Hz input, we achieve more than 88 dB



The resulting USB signal, received in a 60 kHz bandwidth on a Perseus, is very clean….

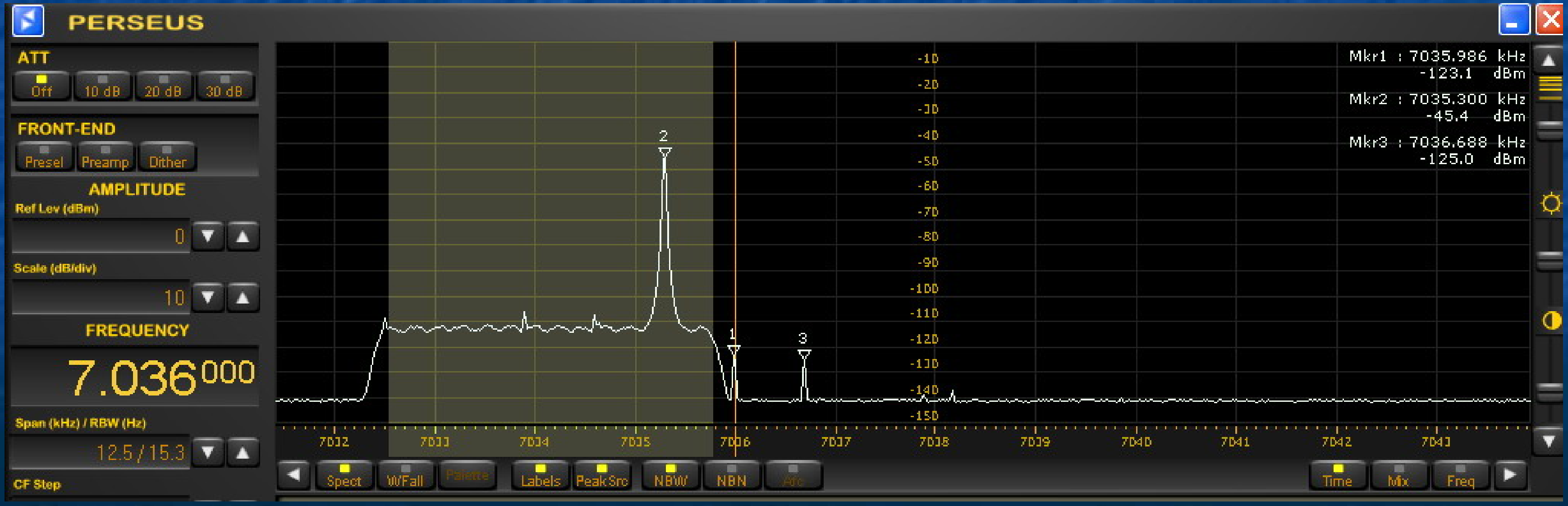


… and likewise in a wider span of 2 MHz.

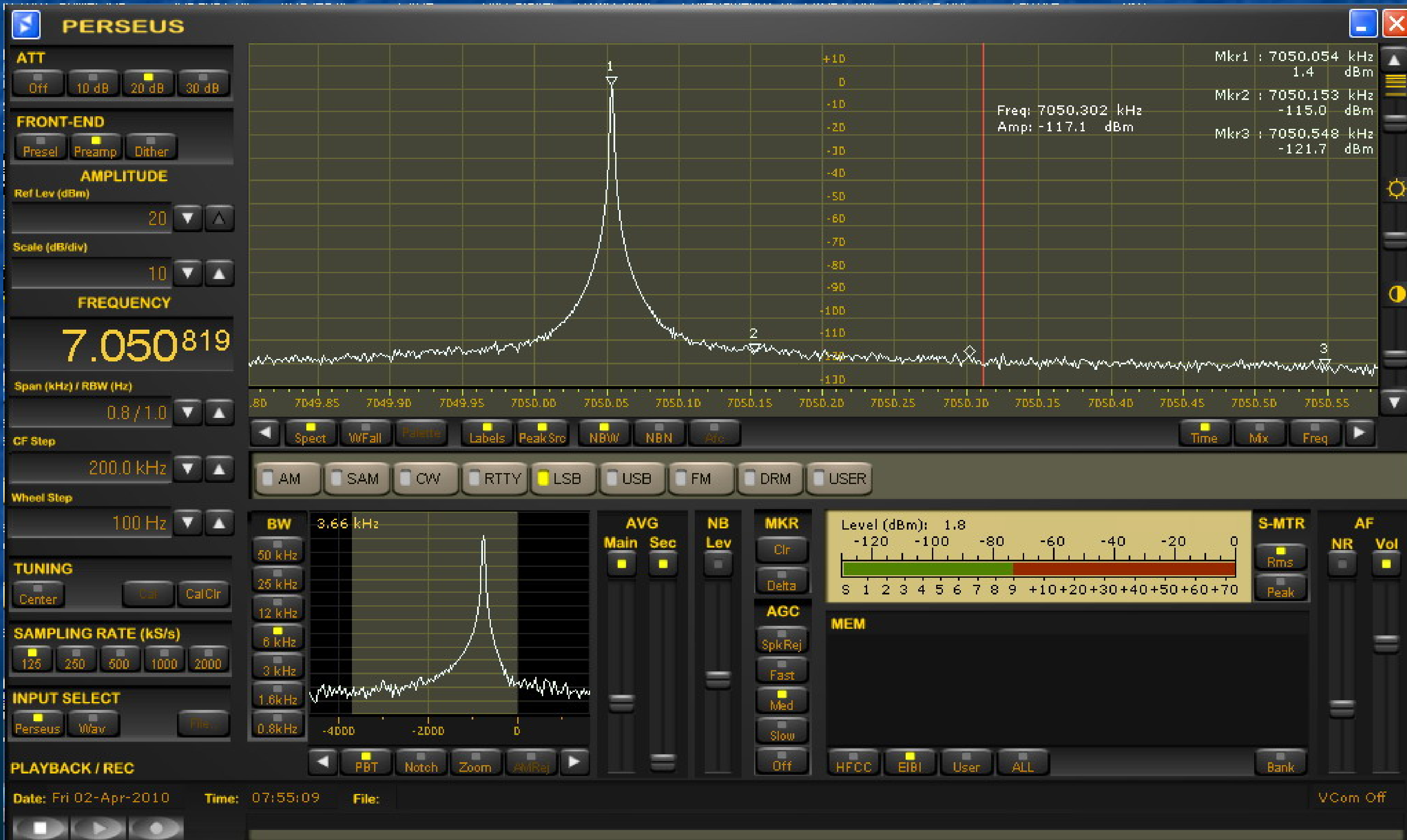


Quality of the generated SSB signal

Carrier attenuation : 79 dB
Unwanted sideband attenuation : 80 dB
2 harmonic attenuation : > 80 dB
3 harmonic attenuation : > 75 dB

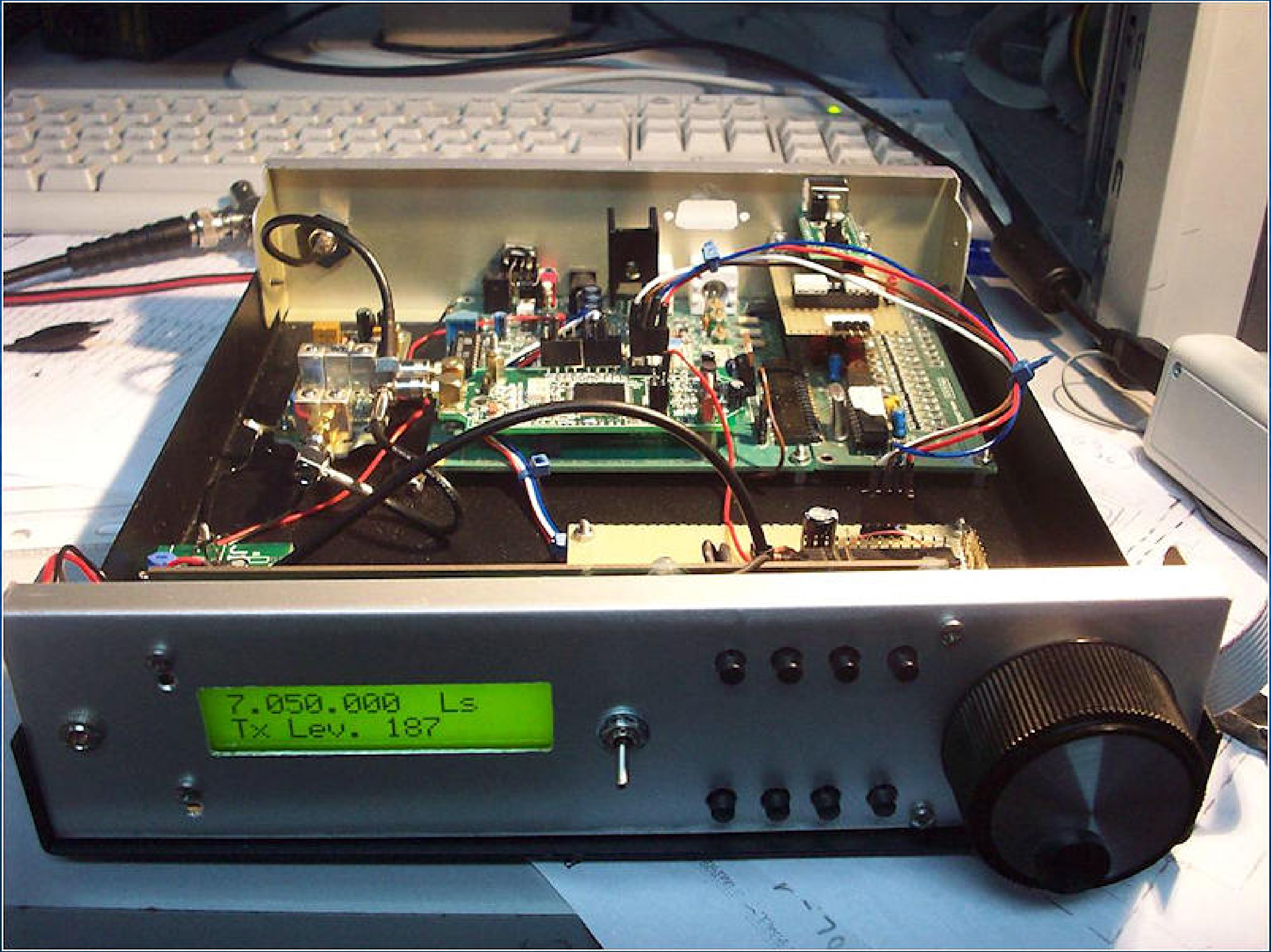


AD9957 phase noise (Single tone mode)

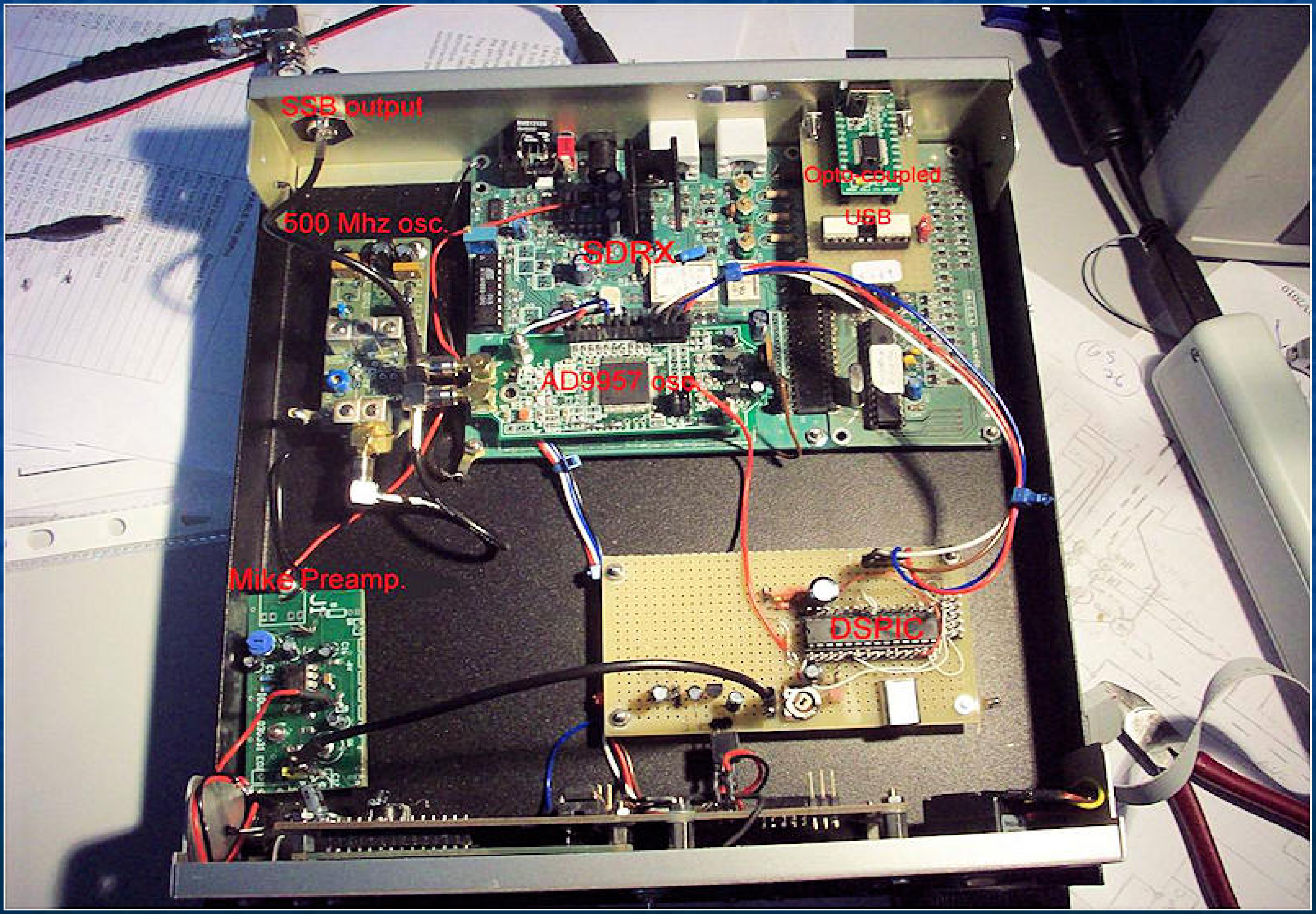


Alberto I2PHD

First prototype produced by Giuliano I0CG



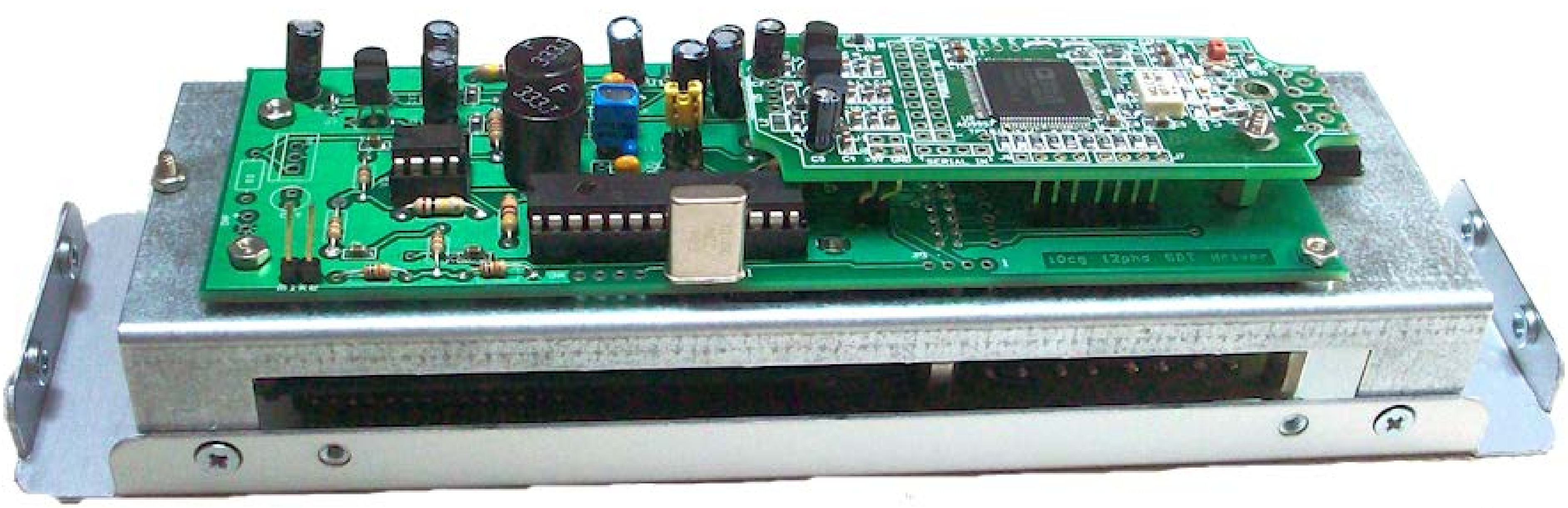
First prototype produced by Giuliano I0CG



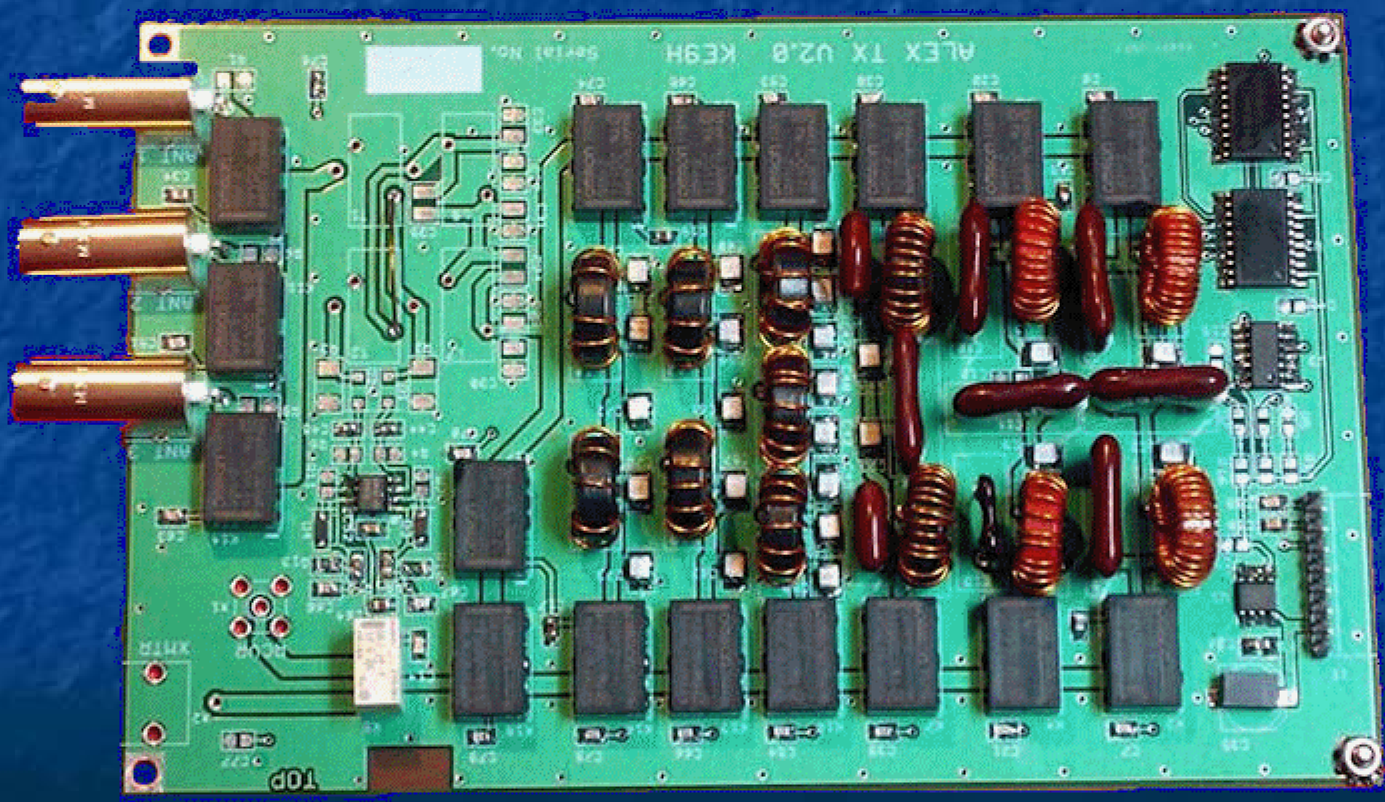
The SSB driver card in the photo (the work of Giuliano I0CG) contains:

1) **Microphone preamp**
2) **Compressor + noise gate**
3) **Anti-alias filter cutting off at 3.5 Khz**
4) **dsPIC**

On top is the AD9957 auxiliary board with RF SSB output at 0 dBm.
The metal work below the PCB houses the front panel.
It does not need any cabling.



It uses a clone of the Alex HPSDR board for the TX passband filter



CAT Protocols used

1) Kenwood protocol modified for WINRAD (uses a DLL by I2PHD for SDRX)

2) Kenwood protocol for POWER SDR

3) ICOM protocol for Perseus or original ICOM RX

These are managed by an emulation of a COM serial port via a photo-coupled USB interface

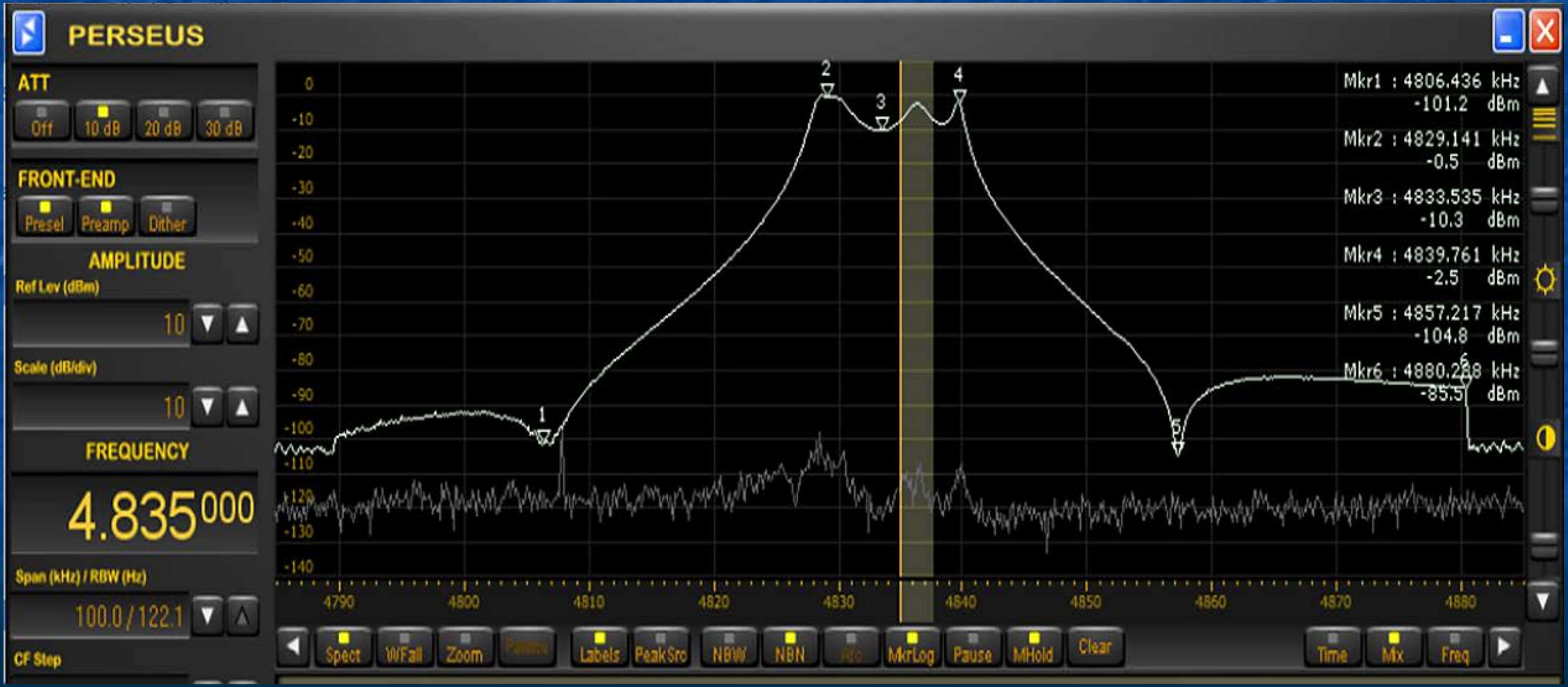
Using the RTX with the Perseus RX

In this case the AD9957 is only used on transmit. On receive, the Perseus does not require a DDS for direct sampling at RF.

So what shall we do with the DDS when we are on receive?

It can be used as a tracking generator for analysing filters or antennas.

Filter analysis using the Perseus RX, with the DDS as a sweep generator



Well, that's it. There are already at least 3 SDTs in use, and when working the USA they get repeated unsolicited reports on the quality of their signals.

See you on the air with Donald !