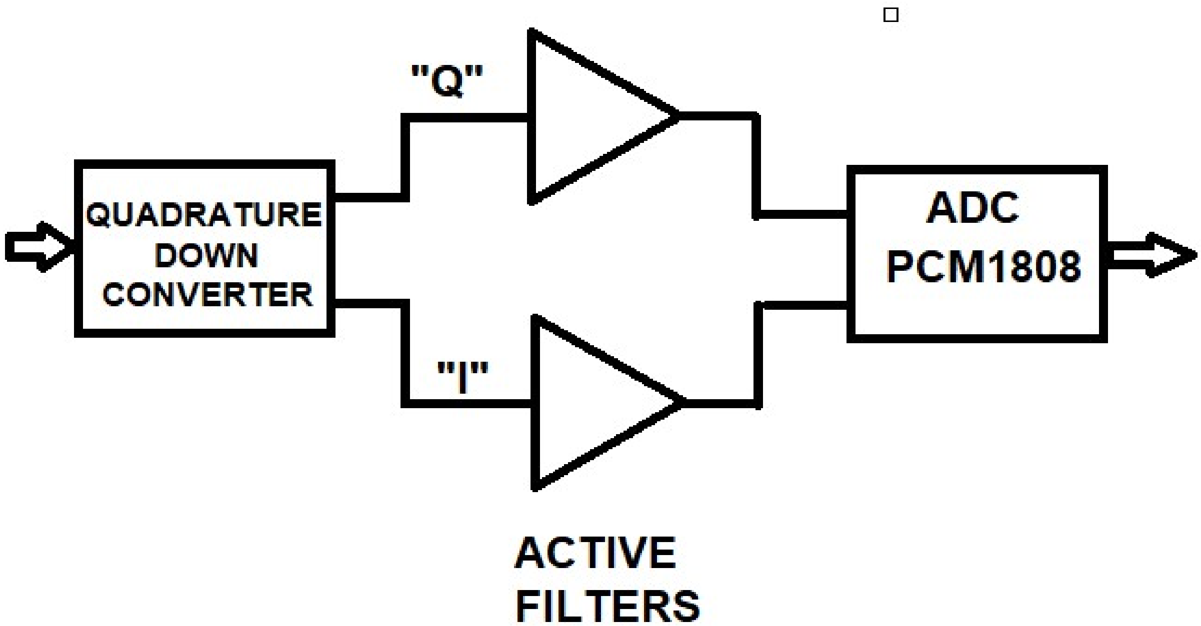
Receive Anti-Aliasing Filters in the T41 By K9HZ (and others noted in the text).

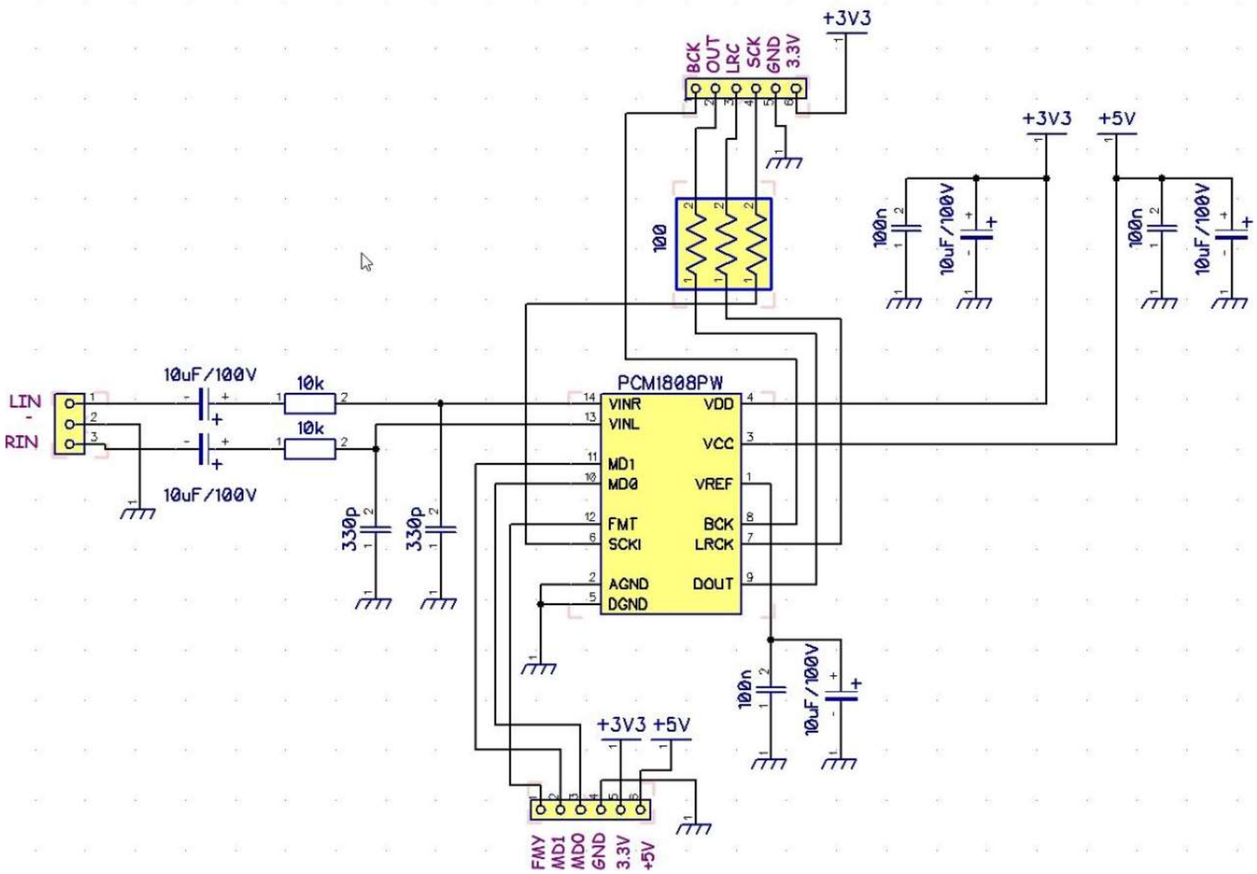
The receiver in the T41 project breaks mixed RF signals into “I” and “Q” components in the audio range for digital processing by the Teensy 4.1. Below is a block diagram of the process:



The PCM1808 is high-performance, low-cost, single-chip, stereo analog-to-digital converter with single-ended analog voltage input. The PCM1808 uses a delta-sigma modulator with 64-times oversampling and includes a digital decimation filter and high-pass filter that removes the dc component of the input signal. The PCM1808 has several sampling frequency selections between 8KHz and 96 KHz. The T41 operates at 96 KHz sampling, but because there are two channels (it is a stereo chip with two independent ADCs), the resolved bandwidth is actually 2 x 96 KHz = 192 KHz.

One important part of the conversion from analog “I” & “Q” signals is the anti-aliasing filtering before processing in the ADCs. Anti-aliasing filtering is required to remove sample data component above the Nyquist sampling theorem limit which would otherwise act as duplicative information (ghosts) in the sampled bandwidth.

As it happens, there is an input filter on the input of the PCM1808 leading to there being two filters in series: the active filters on the QSE board and the passive filter on the PCM1808. The problem here is that the filters on the PCM1808 (each channel) are set to 48 KHz by a 330 pf capacitor and 10K ohm resistor in single order low-pass form:

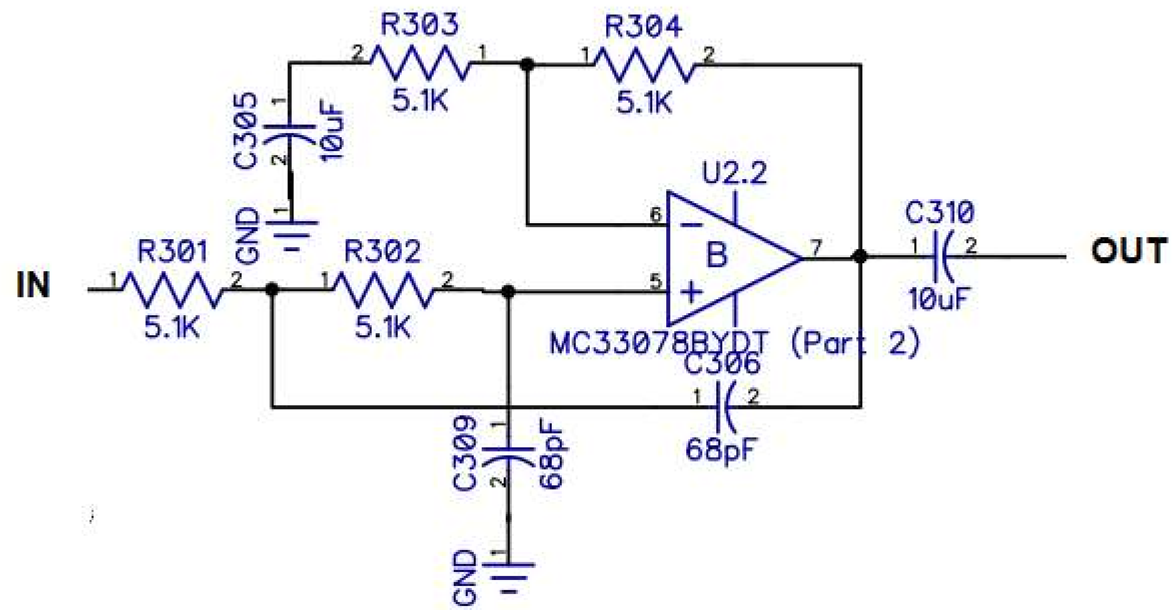


The -3 dB corner frequency point can be calculated from:

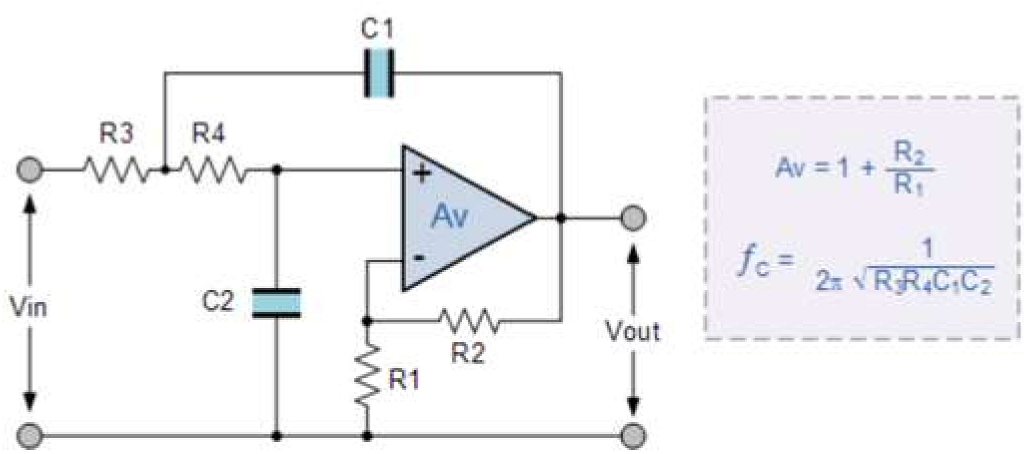
$$F = \frac{1}{2\pi R}$$

Where F is the frequency in cycles per second, R is the resistance in Ohms, and C is the capacitance in Farrads. As originally reported by Alain [callsign], the 330 pf and 10 K ohm filter calculates to a corner frequency of F = 1/(6.28*1e4*3.3e-10) = 48.209 KHz.

The active filter in front of the PCM1808 on the original V11 QSD board is:



This is a second order active filter (generic form):



with a corner frequency calculated by:

$$F = \frac{1}{2\pi\sqrt{R3*R4*C1*C2}}$$

Given the parts that were specified with the V11 build, R3 = R4 = 5100 ohms, C1 = C2 = 68pf. This calculated to a corner frequency of F = 1/(2*3.14*sqrt(5.1e3*5.1e3*6.8e-11*6.8e-11)) = 459 KHz.

The limiting filter in the chain is the one on the PCM1808 and it is unnecessarily restrictive in that the content we are interested in is sampled at 96KHz in each channel. Therefore, it is suggested to REMOVE the 330 pf capacitors on the PCM1808, and replace the 10K ohm resistors with a 0 ohm resistor (short/ wire). This effectively removes the filters from the PCM1808.

Wald, 3Z6AEF also had comments: [T41EP::QSD module::Sallen-Key filters (3z6aef.pl)]

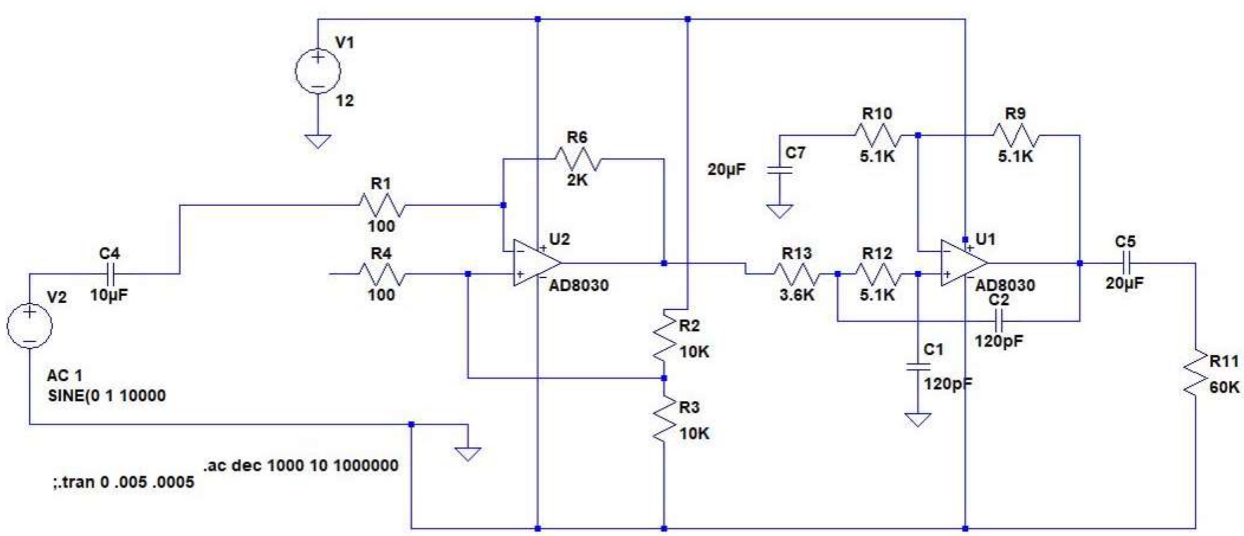
The remaining filter, the active filter on the QSD board, is left to address the anti-aliasing functionality prior to the ADC. Al Peter ran some LTSpice simulations and recommendations to improve the filter:

"To follow up on the discussion, i have implemented the revised anti-alias filter on the QSD board and it works very nicely. I actually don't notice any difference in the operation. The PCM 1808 module also has a very simple anti-aliasing RC filter, which appears to have acut-off frequency somewhere around 300KHz or higher. This does no harm if left in the circuit. However, removing the RC filter does reduce the insertion loss by about 15% or a bit over 1dB. Also, not particularly noticeable. Tried the PCM module w/o the RC filter and it also works just file. Removing the tiny SMDs on the 1808 module is a bit of a pain, as is replacing the 10K series resistor, but it works.

Did a bit of tweaking of the filter and this is a result:

First, the circuit, then the simulated response, and finally the measured response with the new values, using a signal generator input to the Receive module at 1.15MHz,swept over 1 MHz and measured with the scope spectrum analyzer. The two plots are in pretty good agreement. (The simulated response has a log frequency axis and the measured is linear. Both yield better than 30dBreduction at 1MHz and a 3dB down point below 300KHZ.

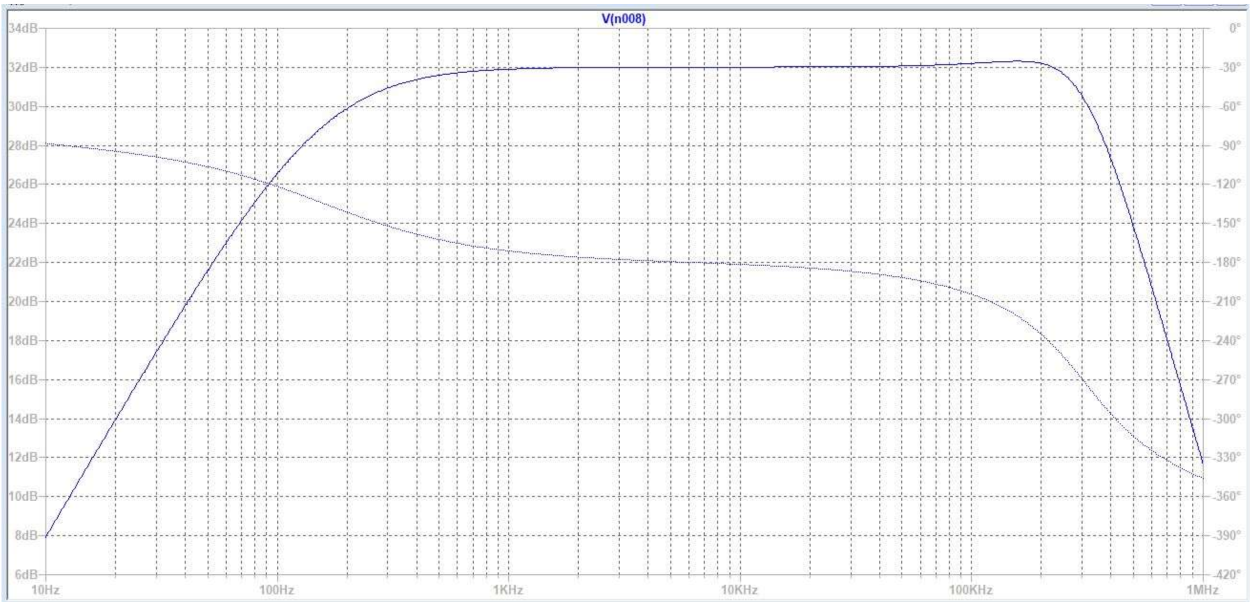
Also sounds good on the air."



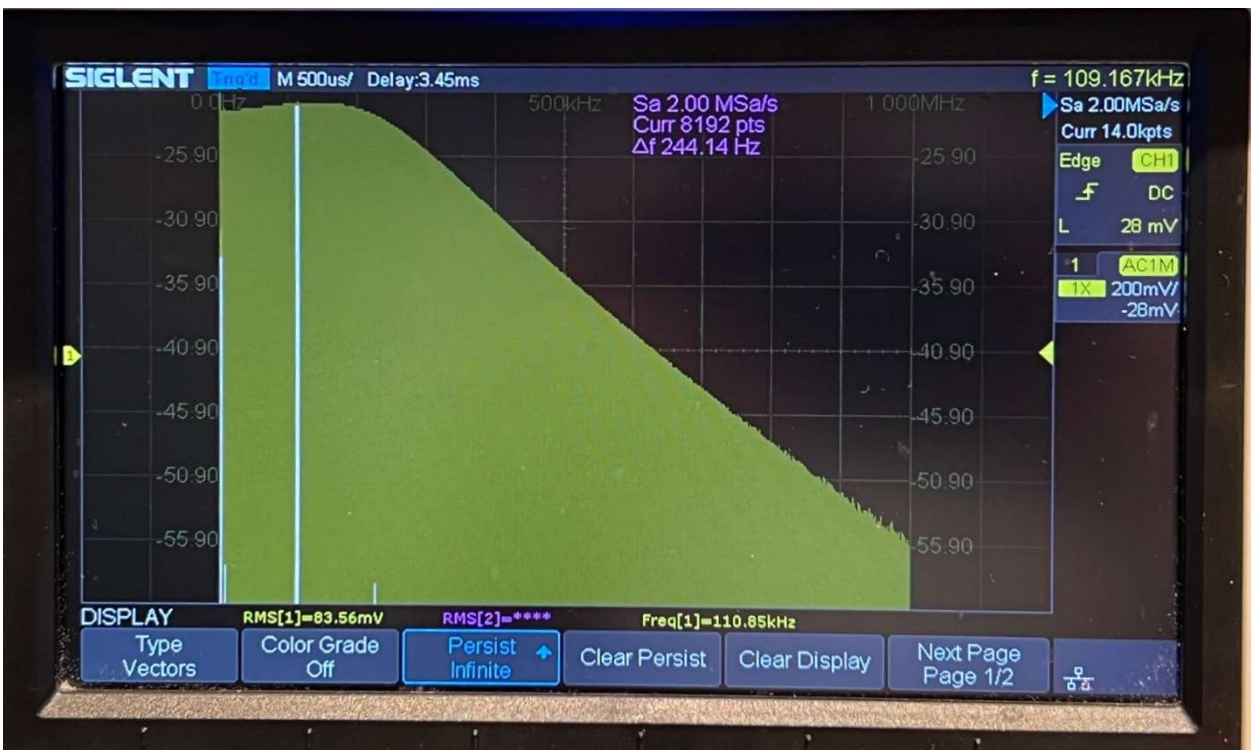
Calculating the corner frequency:

$$F = \frac{1}{2\pi\sqrt{R3*R4*C1*C2}}$$

R3 = 3600 ohms, R4 = 5100 ohms, C1 = C2 = 120 pf. F = 1/(2*3.14*sqrt(3.6e3*5.1e3*1.2e-10*1.2e-10)) = 309.5 KHz. This allows processing of wider-bandwidth signals without loss of information up to the full 96 KHz + 96 Khz signals. Simulation results:



And spectrum analyzer view at the chain:



Note that the filter needs to be revisited if higher sample rate ADC converters are employed in the future.