FUNCubeProPlus - How to use this as the basis of an FM (or AM) Radio Tuner.

The FCPP is a digital tuner/receiver designed to be used via USB connection to a suitable computer. In essence, the combination can then be used as a ‘software defined’ radio tuner. It employs digital methods, but can be used to receive and demodulate ‘analogue’ radio transmissions. This just requires the computer ‘host’ to run a computer routine which is appropriate for the task.

The output from the FCPP is provided via the input being presented to aa digital ‘IQ’ mixer whose output is sampled as a 192k-rate series of sample value pairs. This series of sample pairs can be used to compute the amplitude and phase of the recieved signal at each sampled instant. This can be done by the standard ‘phasor’ approach to processing a series of such IQ sample pairs and recover any information they convey. Here the interest is in being able to receive and analyse/record/listen to (UK) FM broadcast signals. This means being able to recover the details of how the broadcast carrier frequency varies with time as this defines the wanted audio ‘payload’ waveform.

For any IQ sample pair the amplitude, A, at each sampled instant will be given by

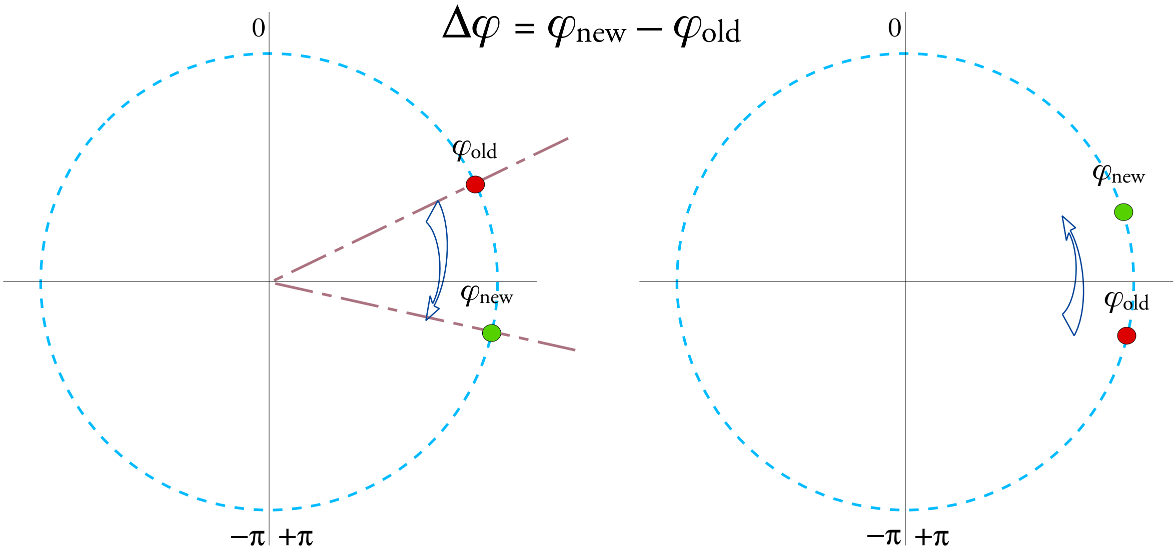
$$A^2 = I^2 + Q^2$$

and the phase of the input signal can be obtained from

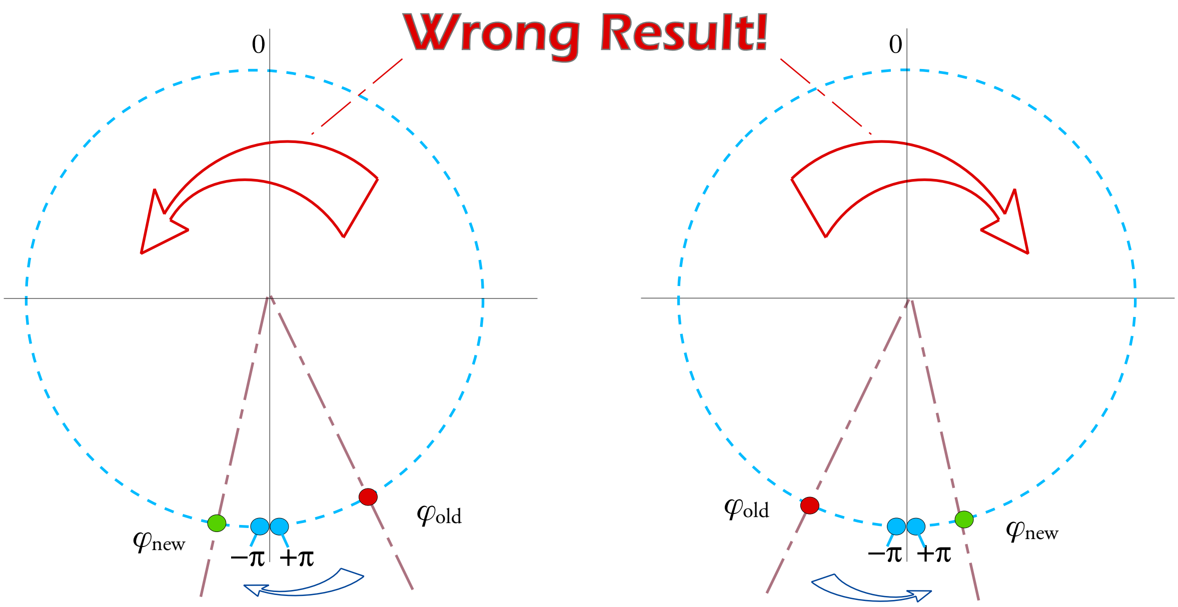
$$\varphi = \mathrm{atan2}\{ Q, I \}$$

For a sinusoid we can say that frequency is, in essence, the rate-of-change of the phase from each sample pair to the next. Hence a series of IQ sample values can be used to determine how both the amplitude and the frequency of the received signal varies with time.

$$\Delta\varphi = \varphi_{\mathrm{new}} - \varphi_{\mathrm{old}}$$



For an FM signal of constant amplitude we can visualise the sample values as being as a series of points around the circumference of a circle, as illustrated above. Two situations are shown. The left-hand diagram represents a situation where the FM signal’s carrier frequency is higher than the frequency of the local oscillator driving the IQ mixer. This means that the series of output IQ value pairs plots out a succession of points going clockwise around a circle (constant signal amplitude). Conversely, when the received frequency is modulated to be below the local oscillator successive sample pairs will go anti-clockwise. This approach enables is to use a series of successive IQ pairs to calculate the FM (and any AM) patterns. However there is one snag with this approach which needs attention. The individual IQ phase measurement values we obtain are all limited to a range between $+\pi$ and $-\pi$.



The above diagram examples this behaviour. For each individual IQ sample pair the phase values we can measure in isolation are limited to the circular range from $-\pi$ to $+\pi$. However successive samples may add contributions that continue to alter the IQ phase *beyond* this range.

This creates a problem due to a discontinuity in the phase circle where "$+\pi$ and $-\pi$ meet"! When this happens using $\Delta\varphi = \varphi_{new} - \varphi_{old}$ no longer gives us the wanted (relatively small) change with the correct sign for the change in phase. Instead it gives us a much larger, *incorrect* value with the wrong sign! To be able to determine the *actual* phase change between successive IQ sample pairs we need to be able to detect and correct for this effect when it occurs.

The nominal range of FM for UK VHF broadcasting is nominally restricted to a peak *Frequency Deviation* of ±75kHz. However in practice the BBC generally tend to aim at a maximum of around ±50kHz. Given an IQ sample rate of 192k sample pairs per sec the time between successive sample pairs will always be 5·2 microseconds. The duration of a half-cycle of 75kHz is 6·6 microseconds, (10 microseconds for 50kHz) and in reality the continuous phase-rate of the audio modulation will be rather lower than this. This indicates that the magnitude of the phase angle change between any two successive samples should always be somewhat less than π. This in turn (pun alert!) allows us to distinguish two quite different cases:

- A crossing of the $-\pi\ /\ +\ \pi$ discontinuity, changing the sign of the next sample.
- A crossing of the 0 value which *also* changes the sign of next sample value.

The result is that we have a way to detect such a sign change between samples. That then allows us to deal with the sign change, identify the direction of 'rotation' (i.e. the sign of the modulation frequency), and deal appropriately with each case.

One cycle of 75kHz has a duration of 13·3 microseconds. The sample interval for 192k sample rate is 5.2 microseconds. Hence the maximum expected change in phase between successive IQ samples should be 0.39 of a cycle – i.e. a phase change of 2.45 radians. Hence when we find a $\Delta\varphi$ value which has a larger magnitude than this we can use this to identify an event of this kind. A complete rotation from $-\pi$ to $+\pi$ represents one complete circle of 2π in phase. We can also see that any "wrong way round the circle" values caused by the discontinuity actually covers all the circle 'outside' the portion of *actual* phase change. It also has the wrong sign.

Thus the correction process can operate on the basis of defining a limit value, λ, based on the above and to check each phase difference between successive IQ sample phases to find any which exceed this limit, and then correct those values using the argument that:

- Any value of $\Delta\varphi = \varphi_{new} - \varphi_{old} > \lambda$ (i.e. +ve and too large) should be corrected by subtracting 2π from them.
- Any value of $\Delta\varphi = \varphi_{new} - \varphi_{old} < -\lambda$ (i.e. -ve and too large) should be corrected by adding 2π to them.

That should – in theory, at least! - then give us the real, correct phase change between the samples.

In practice it may be necessary to slightly alter the chosen λ value because of two other factors.

- Noise. This will cause random variations in the phase values in the series. As a result some sample pairs may as a result cause a 'mis-diagnosis' using the above approach.
- Receiver tuning offset. The receiving system being used to capture a broadcast as a series of IQ values may be 'tuned' to slightly the wrong frequency for the nominal carrier of the FM transmission.

Given high input SNR the noise is unlikely to cause problems. And a slightly increased choice of λ may be sufficient to suppress this – abeit with the risk of then faiing to detect some cases of an error caused by the $-\pi / + \pi$ discontinuity!

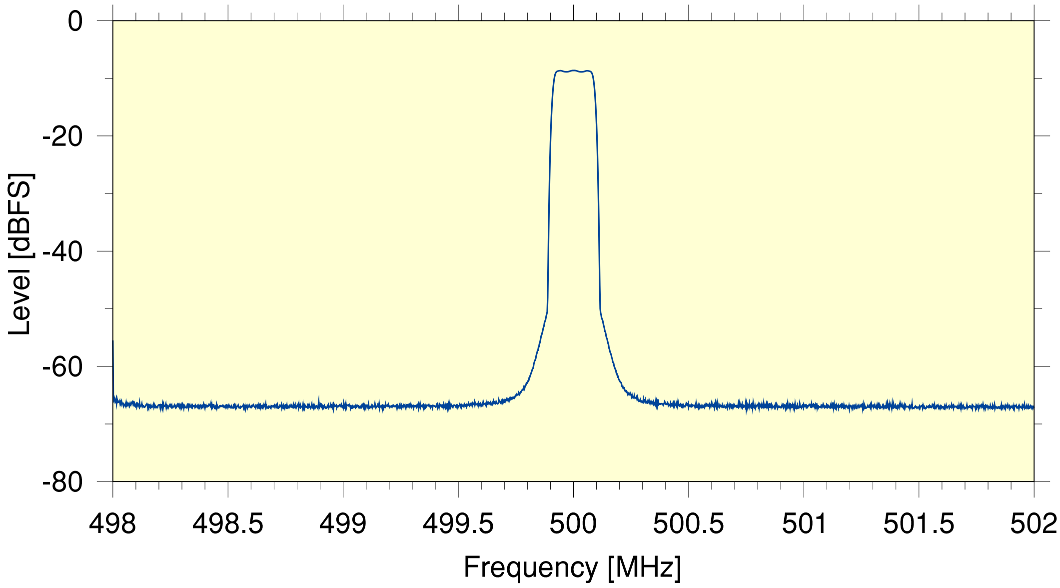
The second factor may cause problems if the IQ receiver is using a slightly *incorrect* frequency to drive its IQ mixer and generate the series of IQ sample values. It is then tuned incorrectly and this may distort the resulting output. e.g. if you set the Local Oscillator to be, say, 10 kHz too low this will cause two problems.

1) The output series of frequency versus time values will exhibit a corresponding DC offset. i.e. all the measured phase-rate value will have an added offset determined by the tuning error. This can be corrected simply by having a DC filter or - in the signal processing software – detect the unwanted DC and use this to determine a fixed value to subtract from each measured phase rate value to remove the offset from the output.

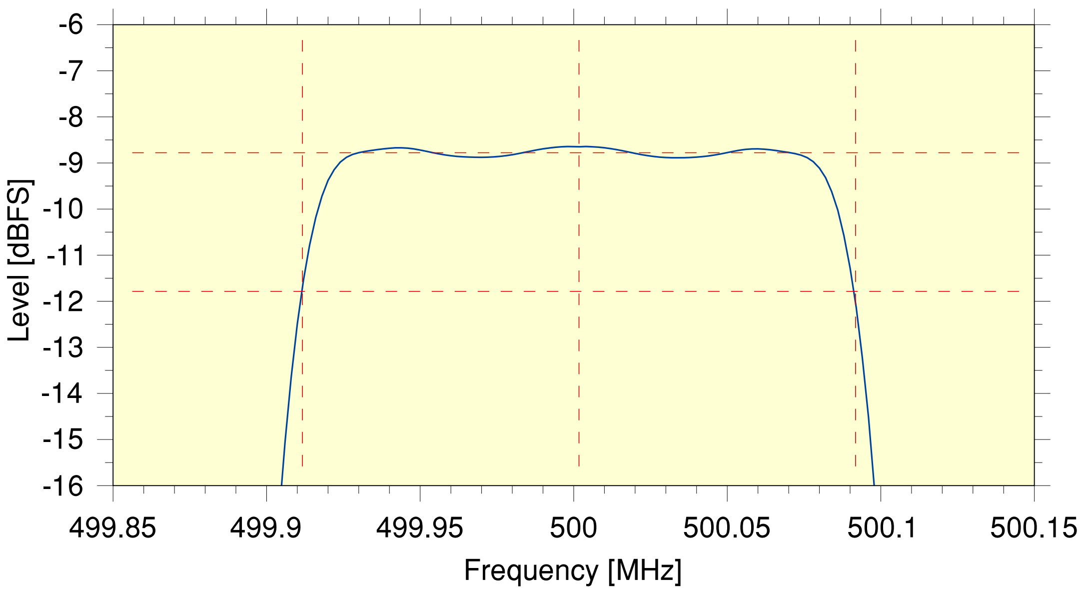
2)The FCPP samples IQ pairs at 192kHz So should in principle be able to cope with tuning frequency errors up to just under 96k - 75k = 21kHz without reaching the limit set by the IQ sampling rate. However if the tuning error is greater than ±21kHz the FM deviations of the broadcast may give a distorted result because the values now exceed the bandwidth of reliable sampling.

FWIW I used QTHID (Linux) to control the tuned frequency, and using the FCPP with an FFT spectrum analyser showing its output was able to tune into FM stations with an apparent accuracy of 1kHz or better. Hence tuning error was not a problem during my tests. More generally an adaptive FM tuner design could easily detect any small dc offset in the output demodulated stream and use that as an automatic frequency control (AFC) to tweak the LO frequency driving the IQ mixer/sampler to deal with this problem. Alternatively, for small dc offsets, it could subtract the determined offset from the values of the output modulation-value samples. Thus one way or another, the best approach is – as far as possible - to ensure accurate tuning of the receiver!

The choice of a finite sampling rate (in this case 192k IQ) comes with a specific limitation imposed by Information Theory. The conversions will only give unabiguously correct phase-rate values for input in the 192kHz range around the unmodulated carrier frequency. Input at frequencies which are beyond the channel we have tuner to may affect the samples and cause interference! Hence it is important to ensure that the IQ sampling process is protected from picking up unwanted interference from outwith the chosen broadcast's channel!



When calibrating the FCPP I also did some scans to check its ability to limit the range of frequencies it would accept. The above shows a typical result. The graph below 'zooms in' to examine the region around the tuned frequency (500 MHz in this case).



Taken together, the two plots show that the FCPP employs filtering which allows it to receive over a range covering a 200 kHz range (at its -3dB points) whilst rejecting unput at other frequencies. This then helps to prevent ‘interference’ from adjacent channels and thus reject the above problem.

Overall, therefore, the FCPP seems quite suitable as a device for use as a ‘digital’ FM Stereo Audio Tuner. The rest would be software… and a decent stereo system to play the results!

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