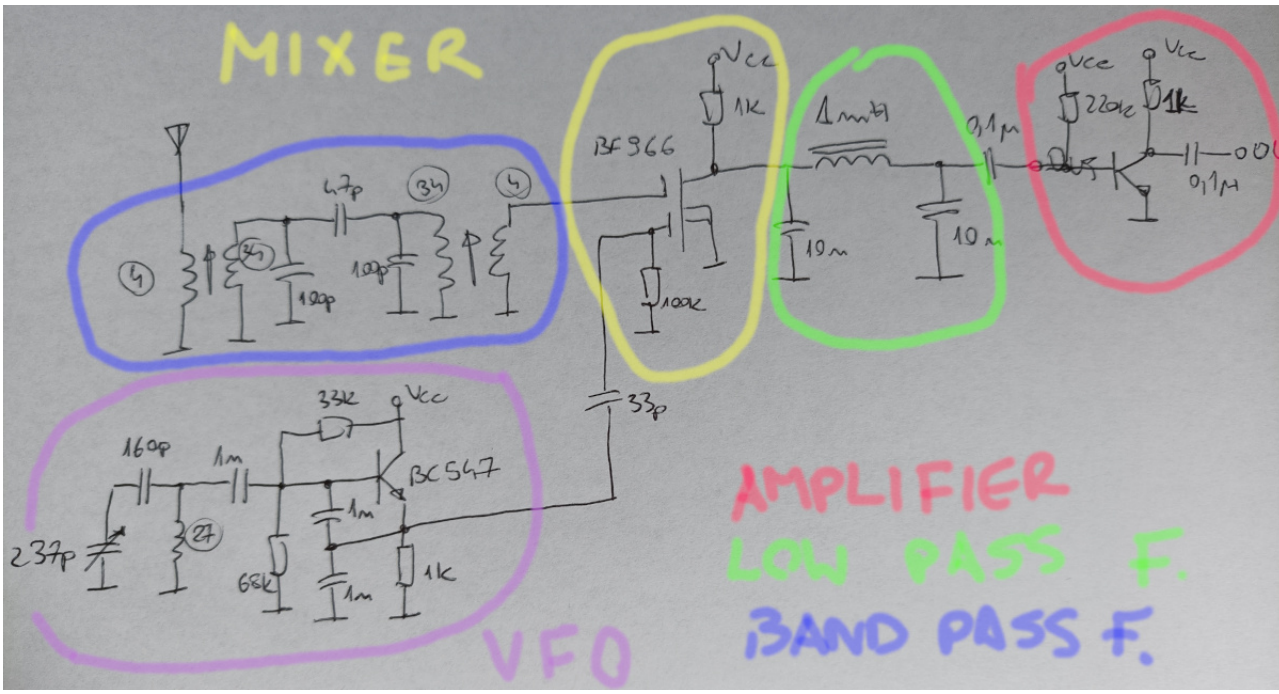
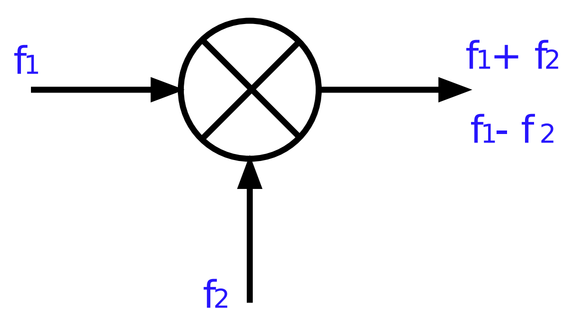
In one of the threads posted on X, I published a hand-drawn diagram of something that could roughly be called an SDR receiver. Since I see that there are people who would like to get acquainted with the principle of operation of this scheme, I decided to write a little more.



Signal mixing

The signal mixing technique is one of the basic principles on which radio technology is based. If you understand the principle of operation of the mixer, it will become clear to you how the circuit in the diagram works. Not only this one, but also many others (they don't differ that much in principles).



Sketch 1

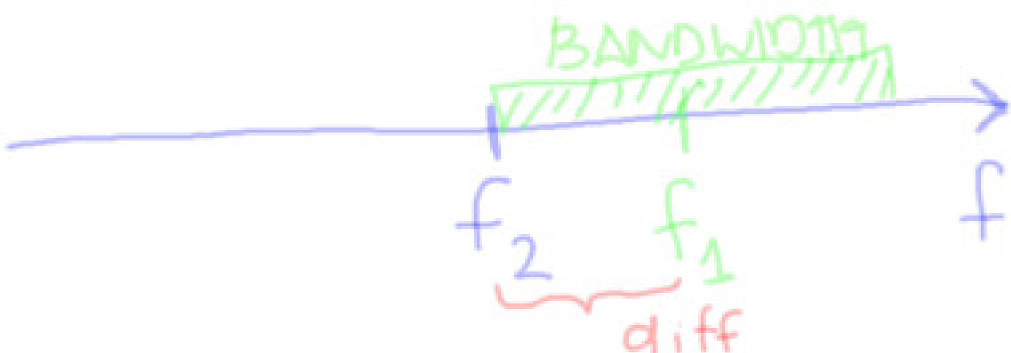
The sketch above shows the mixer symbol [1].

The mixing process combines frequencies f_1 with f_2. As a result of mixing, we obtain two variants of combining these frequencies::

- $f_1 + f_2$
- f_1 - f_2

What is this needed for? It will be best to show this with an example. Let's assume that we want to watch the signal spectrum (a popular waterfall well known to you from SDR applications) at a frequency of 100MHz. In this case, f_1 will be the electromagnetic signal received from the antenna. The frequency f_2 will be the signal supplied from the local VFO (variable frequency oscillator, slightly below the value of f_1), which will operate at the frequency f_2 = (100MHz - diff). What do we get in the mixing process? Let's assign the values into equations:

- $f_1 + f_2 = 100^{\text{diff}}$ MHz+100 MHz=200MHz. That is what we don't need and we filter this one out with a Low Pass Filter (marked green on schematic)
- $f_1 - f_2 = 100^{\text{diff}}$MHz -100 MHz= radio signal transformed to an audio spectrum (marked green below)



Sketch 2

In other words, by the process of mixing frequencies f_1 and f_2, we shifted the electromagnetic signal received from antenna to an audio signal, that can be fed into the audio card, translated into samples and using DSP processing converted into a waterfall plot.

That's basically it!

[1] More info at https://www.electronics-notes.com/articles/radio/rf-mixer/rf-mixing-basics.php

The devil is in the details

Band pass filter

The theory looks quite simple, but in practice our design needs filtering. First of all, we need to filter our signal from the antenna that is fed to the mixer. If we don't, strong signals from completely different frequencies will interfere with the operation of the mixer. Therefore, radio receivers are usually equipped with a set of bandpass filters at the input on that occasion. This is also in our case - it is a BPF filter (marked blue). Such filters come in the form of ready-made modules with band-pass set of filters, switched using relays on the receiver's antenna path. In our solution from the diagram we are dealing with an 80m band pass filter. In the circles I've marked the number of turns of inductive elements that I wound on tunable rf inductors (34 turns should give around 17μH).

Low pass filter

As you have already noticed, as a result of mixing we obtain two signals - one moved to the low-frequency domain, and the other, which is at a frequency approximately twice as high as the received frequency. Therefore, they need to be separated from each other, and this is done using a low-pass filter.

The filter presented in the diagram limits the bandwidth (see Sketch 2) to the range of acoustic frequencies (for the sound card use). In this case, the width of the spectrum we receive is no greater than 20kHz and it results from the limitations of the audio card (Shannon's theorem). Therefore, in this case we receive a signal from 100 MHz to 100.025 MHz. However, if we have sampled our audio signal with a much faster ADC than the one on the generic sound card, we could obtain a value of the order of 100MHz - 120MHz, and then our low-pass filter should be of a different design and operate in the range of up to 20MHz.

Modulation bandwidth

The amount of the received bandwidth is very important. Depending on the type of modulation of the transmitted signal, it occupies a different bandwidth. Using a sound card, we can easily receive CW, AM, SSB, NFM modulation. However, the problem will be receiving an FM radio station (WFM mcodulation), which already occupies 38kHz and - as noted before - a generic sound card will not be capable of handling it.

VFO generator

The schematic diagram shows a generator in the Seiler configuration, operating in the frequency range of 3.5-4MHz. If we wanted to work at higher frequencies - around 100MHz, the values of the capacitors in the base circuit would not be

1nF but approx. 15pF, inductor should be core-less Φ6mm, and have about 6-7 turns of AWG 14. Generators built on LC resonant circuits are very sensitive to the quality of workmanship and environmental conditions. In the event of changes in temperature or humidity (and thus the dielectric permittivity of the air capacitor), our frequency will hunt. Instead of such a generator, you can use the DDS synthesizer AD9833 with SPI interface.