Software Defined Radios (SDR)

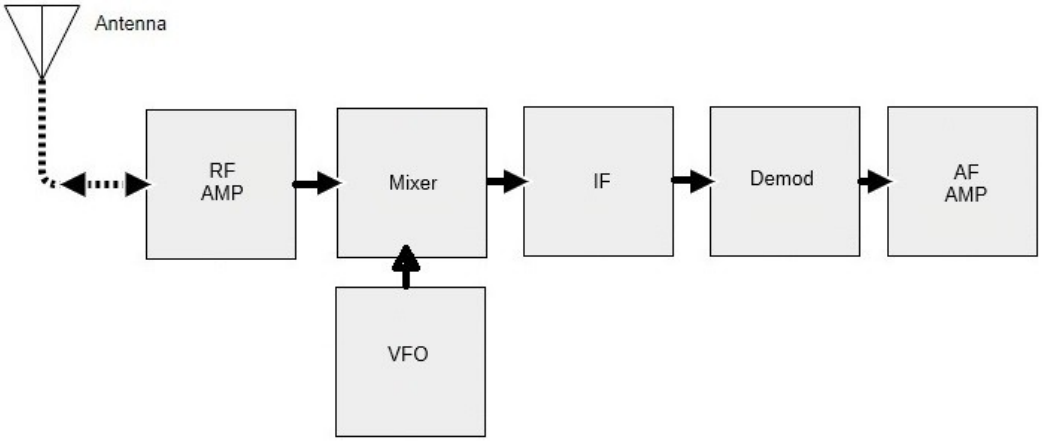
Keith Regli (K7KAR)

In 1960 if you tuned the phone portion of any HF amateur band almost all the stations you'd hear were using traditional AM. By 1970 they were almost all using SSB. The switch from superheterodyne to direct conversion SDR radios is almost as dramatic. A confluence of improvements to certain technologies has made this possible. This discussion will be about the receiver side, but very similar concepts are used on the transmit side.

Remember that a radio signal consists of an electromagnetic wave modulated by an information signal. For the sake of the discussion here we'll take that information signal to be a voice using single sideband modulation and the electromagnetic wave has a frequency of 14.250 MHz

The flow through a Superheterodyne Receiver

Most ham receivers have used this flow since the 1940s. The concept was invented around 1918 by Edwin Armstrong (and others). Better receivers incorporate two or even three IF stages. The description below is a somewhat idealized version of the major functions.



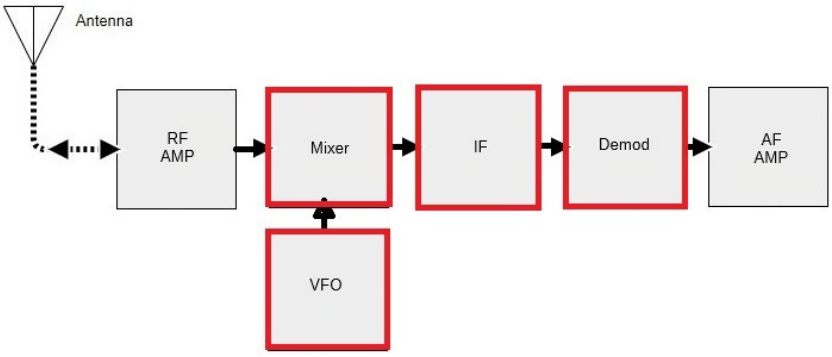
Imagine some ham in Australia is calling CQ on 14.250 MHz. A little bit of his signal is arriving at your antenna along with signals from other hams, AM Broadcast, FM Broadcast, TV stations, Fire, Police, and thousands of other sources. All of these signals excite your antenna which produces (at any instant) a tiny voltage at the feedpoint that is the sum of all the individual signals. The RF Amp attenuates most out-of-band signals and amplifies the in-band signals (here 20 meters). The VFO, controlled by your tuning knob, injects a 5.250 MHz signal to be mixed with the 14 MHz signal(s) from the RF Amp producing four outputs: 5.250, 14.250, 19.500 and 9.000 MHz. The IF stage has a very narrow filter (around 3 kHz) at 9.000 MHz that is able to select just one signal (our Australian). This signal is fed to the demodulator which retrieves the audio information from the 9 MHz signal and sends it to the audio amplifier for us to hear.

One limiting factor in this model is our ability to make a sufficiently sharp filter at the IF frequency. Double and triple conversion variants allow multiple levels of filters and at lower frequencies where we can make a better filter.

How can we do better?

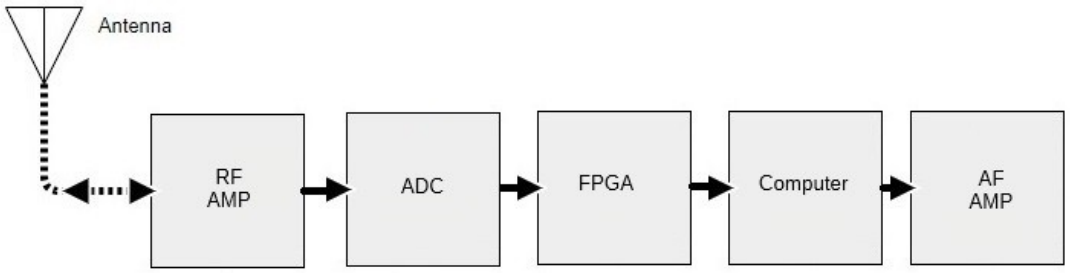
A lot of research has gone into Digital Signal Processing (DSP) in the last few years as Analog to Digital Converters (ADC) have become faster with higher resolutions; making digital representations of higher frequency signals in real time possible. The work of Nyquist and Shannon in the early to middle 1900s informs us as to how fast they need to be to represent a given frequency. The work of Cooley and Tukey on the Fast Fourier Transform (FFT) around 1960 coupled with the work of Wilkinson on ADCs first made possible very sharp, flexible audio filters. ADCs became fast enough and cheap enough that in the early 1900s folks started making direct conversion radios using DSP techniques to replace the IF stage. In a similar timeframe, Bedrosian showed how to use a digital version of the Hilbert Transform to create a single sideband signal from a digital representation of an audio signal. All of these advances allow one to replace much of the dedicated dedicated circuitry from the superheterodyne model and get much better results.

The current state of the art replaces everything but the RF and AF amplifiers by something.



The flow through a Software Defined Receiver

The tiny signals that enter the RF stage are either attenuated or amplified depending on whether they are in the selected passband.



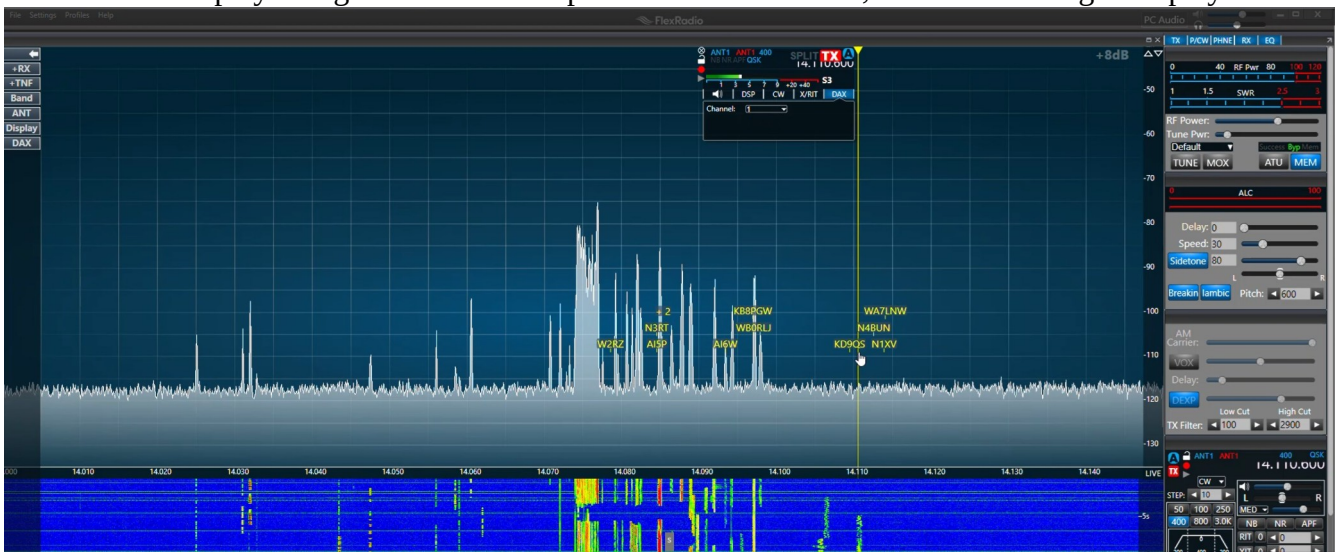
The result is fed to the Analog to Digital Converter (ADC). This very specialized computer takes voltage readings about 250 million times a second and converts each reading to a 16 bit representation of the voltage. Each of those readings is the sum of all the signals at that moment in time. The collection of these readings over time is a digital representation of the original waveform. The result is a firehose of readings sent to the FPGA stage.

The Field Programmable Gate Array (FPGA) is a computer that is very good at doing some very specialized mathematical operations. It uses the Fast Fourier Transform and other algorithms to produce a list that consists of representations of frequencies and amplitudes that make up the signals that came through the RF stage. This list (the IQ data) is constantly being updated based on the latest samples coming out of the ADC.

A generic computer can read this list and use it to display a graph like we saw from SmartSDR. It can also pick out a frequency and see which signals in the list are at (near) that frequency. It can also extract the information necessary to demodulate the selected signals and send those results to the AF stage (which is normally the computer sound card).

Using the SDR model we get a some new capabilities. Using software DSP tools we can make sharper and more flexible filters making more selective receivers. Since most of the controls are in software we can upgrade capabilities easily. Since all of the back-end work is done in a computer we can physically separate the radio and the control (you can have your radio and antenna on a mountain top and your laptop in your living room).

Below is a picture of the software that comes with Flex radios. You can see the controls on the right side and the display of signals from the output of the FPGA. Also, notice the callsigns displayed on the



screen. These are produced by third-party software that connects to the radio/display separately from the SDR software. It either interprets the signals the get the callsigns or pulls them from DX Clusters.

Below you can see several SDRs that are commercially available. The first is only a receiver and the one on the bottom left has the computer built inside the box using a traditional look with knobs and switches.

$30

$350

$2500

$450

$3500

Timelines

It’s interesting to look at some of the history associated with the advent of Software Defined Radios.

1680 Newton and Leibnitz investigate approximating curves (waveforms) by straight lines.

1715 Taylor and MacLaurin investigate approximations with polynomials.

1810 Fourier investigates approximations with sine waves.

1910 Hilbert studies convolution theory producing the Hilbert Transform.

Armstrong produces superheterodyne receiver.

1928 Nyquist studies sampling theory to digitize waveforms.

1945 Shannon studies sample rates necessary to accurately represent waveforms.

1950 Wilkinson creates first analog to digital converter.

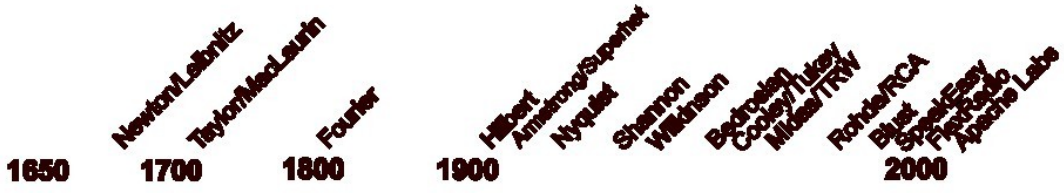
1962 Bedrosian extends Hilbert transform to discrete environment with application to DSP.

1965 Cooley/Tukey produce Fast Fourier Transform extending Fouriers work to discrete environment.

1970 Midas SDR receivers

1982 Rohde SDR radio

1990 DARPA SpeakEasy project evolving SDR.

2002 Flex Radio is formed to produce SDRs.

2005 Apache Labs is formed and produces SDRs.