An introduction to HF Software Defined Radio part 2 – the QSD method

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Designing and producing high performance amateur radio receivers and transceivers is an exceptionally difficult task. Every amplifier stage in a receiver adds noise to the signal and this degrades the noise figure. Non-linear devices like mixers cause intermodulation distortion and local oscillators can add noise and harmonic related signals. The radio manufacturers have built on many years of experience and work very hard to minimise these effects. There is fierce competition among the 'big three' to produce transceivers with excellent receiver performance. Receiver IF frequencies are chosen to reduce the image frequencies and place them outside the ham bands. New receivers have very good local oscillators with low phase noise. Some new receivers have roofing filters, variable pre-selectors, or external microprocessor controlled Hi-Q front end filters to make the receiver selectivity track the wanted receive signal rather than just having a wide filter covering the whole band.

A large part of the reason that SDR receivers have very good receiver performance is that the SDR design eliminates the need for multiple mixers, local oscillators and IF amplifiers. Eliminating these receiver components removes the noise and distortion that they cause. All of the 1st generation 'soundcard' and 2nd generation (A/D conversion in the radio) SDR receivers use a QSD (quadrature sampling detector) which acts like a direct conversion mixer. In most cases they use a circuit known as a Tayloe Detector which was patented by Dan Tayloe N7VE in 2001. The same design can be used in reverse as a quadrature sampling exciter (QSE) to make a signal that can be amplified to create a transmit signal. Gerald Youngblood AC5OG, the CEO of Flex Radio stated in a March 2003 QEX article that the Tayloe detector is the same in concept as modulator designs published by D. H. van Graas, PAØDEN, in 1990 and by Phil Rice, VK3BKR in 1998. However he also states that, "*Traditional commutating mixers do not have capacitors (or integrators) on their output. The capacitor converts the commutating switch from a mixer into a sampling detector (more accurately a track-and hold)".* Anyway Dan holds the patent and most if not all ham radio QSD based receivers use the Tayloe design.

In any discussion about software defined radio it is not long before I and Q streams are mentioned. It is at this point that people's eyes glaze over and they decide that SDR is too complicated for them. It is not really all that difficult, so before we get into how the QSD design works I should explain about direct conversion receivers and 'the phasing method'. QSD based SDRs are direct conversion receivers. This is not a new idea. Direct conversion 'phasing' receivers such as the Central Electronics CE-100v were very popular in the mid-1950s until improvements in filter design made superhetrodyne receivers the preferred design.

Direct conversion receivers are like superhetrodyne receivers except that the IF output is directly at audio frequencies extending from 0Hz up to the bandwidth of the receiver. The local oscillator is at the same frequency as the wanted receive frequency, so in the mixer a USB signal extending to 3kHz above the LO frequency becomes an USB audio signal at 0 to 3kHz. This is great except that it is a mixer so it works both above and below the LO frequency. An RF signal 1kHz below the LO frequency cannot become -1kHZ and is reflected into the audio range with a phase change of 180 degrees interfering with the wanted signal. If it was a USB RF signal it becomes a LSB audio signal due to the 180 degree reflection. Traditionally the image frequency problem is managed in one of two ways; either the signals below (or above) the LO frequency are filtered out before the mixer, in the same way that image signals are filtered out before a mixer in a superhet receiver, or the 'phasing method' is used to eliminate the image frequencies. SDR receivers use the phasing method for image frequency cancellation.

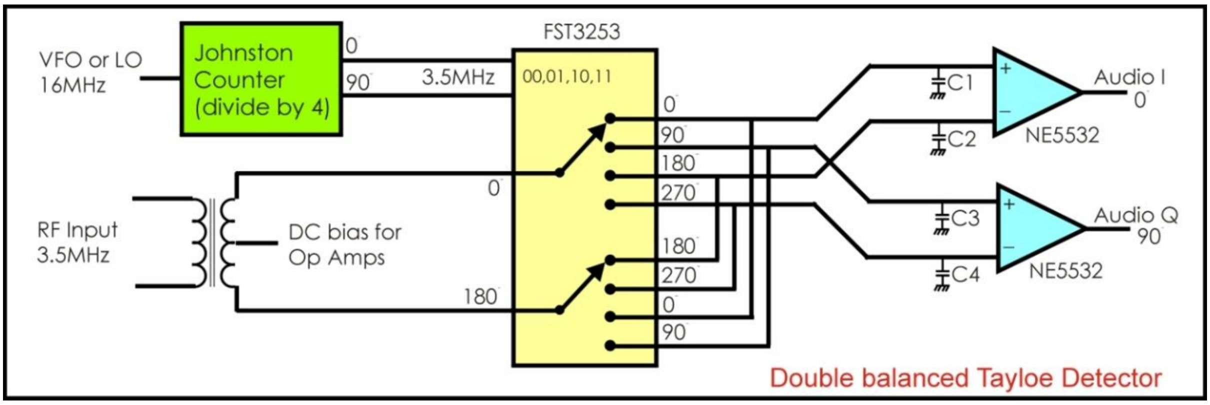
In the phasing method the signal is split into two streams before the mixer and one of them is delayed by 90 degrees. They are called "I" incident and "Q" quadrature signals. After the mixer a 90 degree phase change is applied to the I stream. When the two streams are combined, the wanted signals are back in phase and add together. But the unwanted image signals have a 180 degree phase change so the new 90 degree phase shift causes the two streams to become out of phase and they cancel. The QSD detector or 'switching integrator' creates an audio I stream and an audio Q stream. The Q stream is generated from samples of the input signal taken 90 degrees after the I stream samples. After the detector the signals are converted to digital data using the PC soundcard or a dedicated analog to digital converter (ADC) chip. Software in the PC creates the second phase 90 degree change allowing image cancellation using mathematics rather than circuitry.

The Tayloe QSD detector is very simple and cheap to make. It uses three very basic integrated circuits, a dual Flip Flop latch configured to divide the clock signal by four, a multiplex switch chip and a dual low noise Op-Amp.

Take a look at the schematic block diagram of the slightly more complex 'double balanced' Tayloe detector. The two flip flops in a 74LC74 chip are configured to divide the 14 MHz clock signal by four. The two outputs generate a 00, 01, 10, 11 binary pattern which is used to switch the multiplex switch to each of the four outputs in turn. If the clock VFO signal is from a device like a Si570 chip the receiver can be made to cover a wide range of frequencies. The Tayloe design can be used up to around 1Ghz, but in this example the clock signals to the switch are at 3.5Mhz and the received signals are also centred around 3.5MHz. This means that the signal from one side of the input transformer is switched to all four outputs during every cycle of the input frequency. A pulse of the input signal voltage is applied to each of the four capacitors in turn. The capacitors store the voltage on the inputs to the Op-amps and end up averaging the input levels over time. This works as an envelope detector in the same way that the capacitor following the diode in a crystal set does. In a double balanced Tayloe detector, the signal from the other side of the input transformer is not wasted, because it is anti-phase it is used to top up the capacitor that is 180 degrees out of phase. So for each of the four switch positions two of the capacitors are charged up with the input signal. As Dan states in his article titled "Ultra Low Noise, High Performance, Zero IF Quadrature Product Detector and Preamplifier", "*two separate detectors are driven with the inputs 180 degrees apart using an input transformer. The two detectors use a common set of four detector caps. Since the outputs of the two detectors are 180 degrees out of phase, the detector capacitors are now driven two at a time*". The bias voltage for the Op-Amps is applied to the transformer centre tap and passes through the switch to the Op-Amp inputs.

Because the signal on the 180 degree output is essentially the same as the 0 degree output but with reverse polarity the 0 and 180 degree signals can be combined in the Op-amp. This gives a 6dB improvement in the I stream signal level without adding any noise to the signal. Likewise the 90 degree and 270 degrees are combined in the other Op-Amp to create the Q signal, also with a 6dB enhancement. Just like the 1950s phasing receivers, both of the audio outputs contain the same signals but the Q signal is 90 degrees out of phase. The audio output signals extend from 0Hz up to the bandwidth of the receiver.

The bandwidth of a Tayloe detector is limited by the RC time constant of the capacitors and the overall resistance of the network. Because in the double balanced version the capacitors get topped up twice as often they can be smaller so the bandwidth is wider than the original single balanced version. The other factor limiting the bandwidth is the sample rate of the analog to digital converter following the detector.



Harry Nyquist (1889-1976) determined the fundamental rules for analog to digital conversion. He found that as long as the sample rate was at least twice the rate of the highest frequency in the analog signal, the original analog signal could be recreated accurately from the digital data. Most PC soundcards can do an analog to digital conversion at 48ksps (48,000 samples per second) or 96ksps. Some elite soundcards can manage 192ksps. To keep My Nyquist happy this means that the bandwidth that can be sampled is 24kHz, 48kHz or maybe 96kHz. But because the Tayloe detector is acting as a mixer, each audio stream contains signals from above the LO frequency and signals from below the LO frequency. This means that our SDR receiver can display a spectrum 48kHz or 96kHz wide. If the PC soundcard is a really good one, or a dedicated ADC chip is used you can achieve a bandwidth of 192kHz. The I and Q digital signals can be analysed to determine the amplitude and phase of the signals at the time they were sampled. The phase information is used to decode FM signals and the amplitude information is used to decode amplitude modulated signals like CW, AM and SSB. Two copies of the digital data are created. Using one copy the two signals are compared and a 90 degree phase shift is applied to cancel image signals from below the LO frequency to stop them affecting the signals from above the LO frequency. The resulting wanted signals are displayed on the right side of the spectrum display. At the same time a 270 degree phase change is applied to the other copy and this is used to cancel the cancel image signals from above the LO frequency and stop them affecting the signals from below the LO frequency. The resulting signals are displayed on the left side of the spectrum display. In this way the full spectrum can be shown both above and below the local oscillator or ‘centre frequency’.

A 48kHz or 96kHz bandwidth is enough to display quite a few SSB signals or the entire CW or digital mode section of the band. You can click on any signal and hear the QSO. Another function unique to SDR receivers is that you can record the full bandwidth and play it back later. You can listen to any of the QSOs on the recorded band segment even if you didn’t earlier.

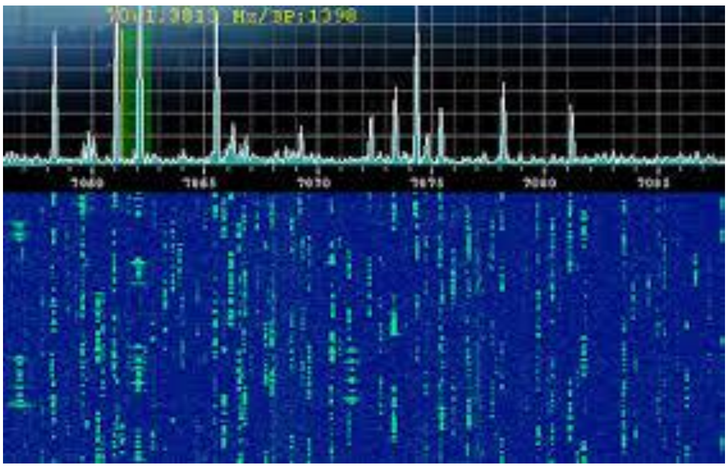
If you have a QSD type receiver you can see the effect of the image cancellation by disconnecting or turning down the level of either the I or the Q stream. You will see that for any signal there is a mirror image on the other side of the spectrum display equidistant from the centre frequency. The mirror images have the sideband reversed which is not a problem and not visible for CW, AM or PSK signals but is obvious with SSB signals.

To get good image cancellation the audio level of the I and Q streams must be the same and the phase difference must be 90 degrees. 40db of image cancellation requires the levels to be within 0.1dB and the phase to be within 1 degree. 60dB requires 0.01dB and 0.1 degrees. The PC software is able to compensate for phase and amplitude errors but it is not perfect across the whole spectrum. The number of bits that the ADC uses to code the data affects the dynamic range of the RF signals that the receiver can handle. The theoretical maximum for a 16bit A/D conversion is 96dB and for a 24bit A/D conversion it improves to 144dB. In the real world you can expect a maximum of around 130dB. Even the 16bit performance is much better than a typical conventional receiver dynamic range of 75 – 80dB, so the SDR does not need an AGC loop to limit strong signals.

Some other statistics for the Tayloe detector include a conversion loss of 0.9dB (typical Rx mixer = 6 to 8dB), a low Noise Figure around 3.9dB (typical HF Rx anything up to 20dB), and a 3rd order intercept point of +30dBm. It also has 6dB of gain without adding noise and it is very cheap.

The QSD SDR has provided very good receiver performance at a lower cost than conventional receivers and it has achieved the goal of making the modulator and demodulator a part of the digital signal processing, bridging the gap between IF DSP and AF DSP processors. But it still has one mixing process which is a potential source of intermodulation distortion. The next logical step is to eliminate the QSD mixer and sample the RF spectrum directly at the antenna. This is the basis of 4th generation SDRs. The process is known as Direct Digital Sampling (DDS).

DDS software defined radios sample the entire HF spectrum from a few kHz up to 55MHz or higher at once. To comply with the Nyquist theorem this requires very fast analog to digital converters which have only become available in the last few years. The next article deals with this new method of implementing an SDR. You can contact me to ask questions at zl3dw@nzart.org.nz or check out my web site at www.qsl.net/zl3dw.



CW signals on an SDR, set the receive bandwidth narrow and just click on any signal. You can almost read the CW signal on the waterfall.