High-performance HF transceiver design: A ham's perspective

By Doug Grant -- EDN, May 3, 2012

The article that appeared in this magazine and in EDN last year about ham radio in the 21st century generated a lot of interest and a lot of questions about various aspects of the hobby. This article grew out of that interest, but is not strictly about ham radio. It is about the design tradeoffs that engineers make every day in designing all kinds of electronic equipment. HF transceivers are used as an illustrative example.

Radio design is a lot like other equipment design

The design of a modern high-performance transceiver for amateur radio is really not much different from the design of many other systems. For example, the volumes are modest, in the few thousands of units per year for a typical model. This precludes the use of ASICs, since most ASICs require much higher volumes to be economically viable. In addition, the moderate volumes don't often attract the product planners of major semiconductor houses.

This means that the design engineer cannot rely on simply marrying a few purpose-built application-specific chips together and call it a hardware design. Nor can he rely on a manufacturer to prepare a ready-made reference design, as is done in high-volume markets like PCs, cellphones, and tablets. Instead, lots of components must be chosen and made to work together to meet the system's performance goals. It takes creativity and use of all available technologies to get a new product into the market, meeting both the performance and cost targets.

Like most systems, compliance to some government regulatory standards is required. In the case of amateur radio transceivers, the government is mainly concerned with the transmitted signal's purity to prevent interference to other services. In the U.S., the applicable rule is 47CFR97.313(d) which states: "... the mean power of any spurious emission from a station transmitter or external RF power amplifier transmitting on a frequency below 30 MHz must be at least 43 dB below the mean power of the fundamental emission." Most amateur transceivers are designed for sale worldwide, and must comply with a variety of other regional standards (CE, for example) before they can be sold.

Beyond the government's regulations, the customers are very sensitive to performance - and will often pay a premium for it. The same could be said about many industrial systems, test and measurement instruments, and medical equipment. In low-performance systems, price becomes the dominant specification, and profits are hard to come by. On the other hand, a focus on high performance can yield good returns.

This article will focus primarily on the design of the receiver side of several modern high-performance amateur transceivers. There are other HF transceivers available, designed for commercial and government application where the customer can afford a higher price than an individual consumer. Delivering high performance while meeting a consumer price point adds additional challenges to the design task.

As with most systems, the first thing to do is define the worst-case system performance requirements. For high-performance receivers, the salient specifications are sensitivity (ability to hear weak signals), selectivity (ability to reject unwanted signals), and various ways of specifying the overall linearity of the signal chain. Linearity is important because any non-linear stages that receive multiple signals (or even one single large signal) will cause artifacts that are indistinguishable from real signals. And like audiophiles, connoisseurs of radio performance (radiophiles?) have certain test cases and on-the-air circumstances that they use to determine a radio's real performance.

Sensitivity is actually the easiest specification to accomplish. It is relatively easy to design a receiver for the HF spectrum (3-30 MHz) with sufficiently low noise figure that the system noise floor is set by atmospheric noise, not receiver noise. The generally-accepted test for an amateur receiver's Minimum Discernible Signal (MDS) is to determine the input RF level that raises the audio output in a narrow bandwidth (500 Hz or so) by 3 dB compared to no input signal. Most modern receivers exhibit MDS on the order of -135 dBm.

Amateur radio is one of the few licensed HF radio services that does not use specified discrete channels (the recently-assigned 60-meter band is the only amateur band that uses fixed channels). In any given band, stations are free to use any frequency that is not in use by another station. Signal spacing can be surprisingly small. Consider the spectrum photo in Figure 1. This shows seven separate CW Morse-code signals in a 2 kHz bandwidth (less than the width of one typical single-sideband voice signal).

Figure 1. The top half of this photo shows seven CW signals in this spectrum analyzer display occupying less bandwidth than one single-sideband voice signal - the bottom half is a scrolling "waterfall" display of the same signals.

Selectivity can be achieved by analog filtering (usually multi-pole crystal lattice filters), DSP, or a combination of the two. The optimum tradeoff of analog/digital filtering, and where in the signal path it is applied, is an ongoing cost/performance compromise that changes with each generation of radio design...just like any other systems.

Linearity is the difficult specification. It determines how well a radio performs in the presence of other nearby signals. Various test methods are used to determine a radio's
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In a receiver with linearity problems, spurious signals can result if the input signals are large enough. For example, the signals at 1823.5 kHz (f1) and 1824.0 kHz (f2) might produce a 3rd-order intermodulation product at 2f1 - f2 or 1823 kHz, well within the passband. No amount of filtering can eliminate this signal once it has been generated, since subsequent stages have no way of differentiating it from a real signal.

The problem is exacerbated when a large number of signals impinge on the front end of the receiver. In Figure 2, a 50-kHz slice of spectrum is shown during a popular operating event.

|Figure 2. This display, captured during a popular on-the-air competition shows over 100 different signals sharing a 50 kHz bandwidth.|

Designing to the Test

Over the past decade or so, independent published reviews of transceiver performance have become increasingly important to the buying decision. There are two primary U.S.-based review sources. The first is QST magazine, published by the American Radio Relay League (ARRL, www.arrl.org). The test methods used in QST Product Reviews have evolved to add new tests from time to time that improve the ability to evaluate radio performance, and include a comparison of the manufacturer's claimed specifications and actual lab measurements.

Another highly-regarded source for product reviews is Sherwood Engineering (www.sherweng.com). The proprietor, Rob Sherwood, has been testing receivers for several decades, and offers aftermarket filter kits to improve receiver performance.

When a new transceiver comes to market, Sherwood conducts a thorough evaluation and adds the measured performance to a table listing nearly every high-end receiver he has ever tested. Radios that test at the top of the Sherwood list (which is sorted by the results of a close-spaced intermodulation test) and the QST review, can make the claim that their receiver is The Highest Performing.

At the time of this writing, the Yaesu FTDX5000D and Elecraft K3 top the list, with the Yaesu radio demonstrating higher Third-order Intercept (IP3) in the ARRL's tests. In fact, both receivers tested so well in the close-spaced dynamic-range test that the measurements were phase-noise-limited (and these radios have low phase noise!).

The reason that these receivers achieve such high numbers, compared to previous generations of high-end radios is driven by the changing test methods and emphasis on a different performance specification. In the past, intermod testing was done at relatively wide signal spacing - typically 20 kHz. This meant that the first IF filter could be 5 kHz or 10 kHz wide, and still do a good job of preventing signals 20 kHz and 40 kHz removed from the desired signal from encountering an amplifier or second mixer that might introduce nonlinearity. Many manufacturers adopted an up-converting architecture with a first IF above the signal frequency, allowing for cheaper, smaller filters, and easier all-band coverage.

However, as the performance tests evolved to use tighter spacing of the two test signals (2 kHz), many receivers using modest filtering at the first (up-converted) IF exhibited quite poor performance, since both signals passed through the wide filter and went through imperfect gain and mixing stages. Some manufacturers reverted to using lower IF frequencies and began to use narrower "roofing" filters that had become available in the first IF. For example, the Yaesu radio includes a 300 Hz filter in the 9 MHz IF stage. Combined with a front-end designed for good linearity in the presence of large signals, the intermodulation dynamic range measurement was outside the ARRL's normal test range and became phase-noise limited.

In the calculated IP3 test, the Yaesu receiver also went off the ARRL's charts with +41 and +40dBm at 20 kHz and 2 kHz, respectively (the Elecraft radio achieved +29 and +28 dBm).

The test method for this "intermodulation dynamic range" is to tune the receiver to the frequency of the expected intermodulation product and introduce the two interferers as carriers spaced 2 kHz apart and increase their amplitude until the audio output level rises 3 dB above the noise floor. The IMD blocking figure is the difference between the input level and the MDS, and is typically close to 100 dB for high-performance receivers.

Another hard-to-meet specification

In a superheterodyne receiver, the incoming RF signal is mixed with a tunable local oscillator (LO) to move the desired signal to the IF stage where it can be filtered and amplified. Local oscillators are imperfect, and produce a signal at the desired frequency plus noise spread out on each side. The mechanisms that create these noise sidebands are well documented and depend on the oscillator design, whether a simple LC-tuned circuit or a design based on a phase-locked-loop (PLL) or direct-digital-synthesis (DDS). In PLLs, the loop bandwidth shapes the phase noise, and is a tradeoff of settling time and step size.

The effect of LO phase noise in a radio receiver is often referred to as "reciprocal mixing noise", and is tested by applying a very pure stable CW signal to the receiver under test, and tuning away from the signal and observing the rise in the noise. In principle, the noise of the LO is down-converted and detected in the same way an input signal with noisy sidebands would be detected if the LO were perfect.

Early receivers used analog L-C oscillators to produce the tunable LO, sometimes combined with a set of fixed-frequency heterodyne oscillators to extend the range. PLL technology implemented with discrete components became popular in the 1970s, with multiple iterations before manufacturers tamed the phase noise.

As DDS became popular in the 1990s, transceiver manufacturers adopted it as a way of generating an LO with lower phase noise. And finally, as both PLL and DDS ICs improved in
performance and integration, novel approaches combining the two have produced excellent results. In order to reach the highest possible performance for this application, most transceiver designers have used integrated PLL and DDS chips but enhanced them with external components and using the chips well below their maximum rated frequencies.

The Ten-Tec Orion series of transceivers uses such a combination of PLL and DDS technologies to achieve extremely high performance. The block diagram is shown in Figure 3. A complete description and detailed schematic of this circuit can be found at https://www.tentec.com/downloads/manuals/566/566_syn_article.pdf.

The LO tunes from 10.8 to 39 MHz in 1 Hz steps for high-side injection to the mixer, producing a 9 MHz IF output when tuning the HF spectrum from 1.8 to 30 MHz. There are actually two PLL loops in operation. A stable TCXO and divider generates two fixed outputs at 44.55 MHz and 7.425 MHz. The 7.425 MHz signal is the reference frequency for a coarse PLL/VCO that generates an output tuned from 549 to 787 MHz in 7.425 MHz steps.

The 44.55 MHz signal is used as the reference for the DDS, which produces an output that is tuned from 2 to 9.425 MHz in 20, 30, 40, or 60 Hz steps and becomes the reference for a phase-locked loop with a VCO that tunes from 541 to 780 MHz in the same 20, 30, 40, or 60 Hz steps. This signal is mixed down using the first PLL's output as an LO, and the downconverted signal is phase locked to the DDS output.

The VCO output is also divided by 20, 30, 40 or 60, yielding the necessary output range and 1 Hz tuning step size for each band. The DDS chip in this circuit is specified up to a 300 MHz output frequency, but is operated at much lower frequencies, since the spurious outputs were judged too high above 10 MHz and would have caused artifacts in the LO circuitry.

This design achieves very low phase noise, on the order of 130dBc/Hz at small frequency offsets. An interesting side effect of this architecture is that the phase noise at any given offset is lower as the division ratio is raised, yielding lower phase noise on the lower bands. This is beneficial because on the lower-frequency bands, signals from distant stations are much weaker relative to local stations than on the higher-frequency bands.

Elecraft uses a different approach to LO generation in their high-end K3 transceiver. Both PLL and DDS technologies are used in the design. Figure 4 shows a block diagram of the K3 synthesizer. The complete schematic can be found at http://www.elecraft.com/manual/K3_Schematics_Jun_2010.pdf.

The master frequency reference is a fixed 49.380 MHz temperature-compensated crystal oscillator (TCXO). A low-power 75 MHz DDS chip is used to produce the reference for the PLL, but the DDS is operated at 8.215 MHz and only adjusted over a narrow range (+/- 2 kHz in 0.2 Hz steps). The output is cleansed of any spurs by a 4-pole crystal filter that is about 1.2 kHz wide. A simple PLL is used for the tunable LO, with a 3-pole active loop filter. A complex discrete VCO design is employed, with a total of 128 different L-C tank circuit combinations used, selected by software depending on the segment of the 8-46 MHz tuning range.

Both approaches result in excellent phase-noise performance as shown in Table 1.
Table 1. Here is a comparison of the reciprocal mixing noise measurements for three high-performance transceivers. Three different designs yield quite different results, all of which are outstanding. (Source: Sherwood Engineering)

<table>
<thead>
<tr>
<th>Offset, kHz</th>
<th>TenTec Orion 2 20M</th>
<th>TenTec Orion 2 40M</th>
<th>Elecraft K3 20M</th>
<th>Yaesu FTDX5000 20M</th>
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</table>

The K3 transceiver is a modular design, and one option is to add a complete second receiver chain, or sub-receiver. The sub-receiver can be used to listen on another frequency, or to listen on the same frequency as the main receiver, but with a separate receiving antenna favoring a different direction in an effort to reduce interference or fading. This second option is called diversity reception. In some transceivers the subreceiver is lower in performance and offers a limited version of diversity. In the K3, the subreceiver is identical to the main receiver, and LO is phase-locked to the main receiver LO, yielding true diversity reception.

Both the TenTec Orion series and the K3 use DSP for the detection and final receiver filtering. In the Orion, the first IF is at 9 MHz including narrow roofing filters, a second IF is at 455 kHz, and a third IF at 14 kHz is delivered to a 24-bit A/D converter designed for high-performance audio system. In the K3, the output of the first IF stage at 8.215 MHz is mixed with a fixed 8.230 MHz LO and down-converted to a 15 kHz second IF. The signal here is digitized using a 24-bit A/D converter, also originally developed for high-performance audio systems.

And now for something completely different

Microtelecom took a different approach to receiver design in the Perseus software-defined receiver. This might be considered the brute-force digitizing method. The Perseus receiver features an extremely simple block diagram as shown in Figure 5 - a front-end passive attenuator and filter, and an amplifier stage connected to a very wideband A/D converter, followed by a programmable digital down-converter (DDC) and USB interface.

Figure 5. The Perseus receiver by microtelecom uses a direct RF-sampling A/D converter and SDR architecture to eliminate analog receiver imperfections, but requires a PC and software for signal demodulation.

The DDC, implemented in an FPGA, includes a numerically-controlled local oscillator (all-digital) and a quadrature digital mixer that delivers baseband I/Q outputs at 100, 200, or 400 kps rates to a USB interface. PC software controls the receiver and performs signal-detection and demodulation functions in the PC's processor. The performance of this complete system compares favorably with traditional all-analog and hybrid analog/DSP designs. For example, LO phase noise is virtually eliminated, since there is no LO! The only source of reciprocal mixing noise is sampling jitter in the A/D converter and phase noise in the clock oscillator, which is a fixed-frequency high-stability crystal oscillator and quite clean.

The astute reader will notice that unlike the 24-bit A/D converters used on the TenTec and Elecraft radios, the Perseus only uses 14 bits of A/D converter resolution. Since a 14-bit A/D converter can only provide about 86 dB of theoretical S/N ratio (and in fact the LTC2206-14A/D converter used only provides about 77 dB SNR) this would suggest that the available dynamic range of the Perseus would be much poorer than the other radios with higher A/D resolution.

However, consider that this noise, including quantization effects, and all other noise sources, is more-or-less spread out evenly across the entire bandwidth of this 80 MSPS A/D converter from dc to Fs/2, or 40 MHz. If we are only concerned with the amount of a small slice of that bandwidth, say, 500 Hz, and remove all the other noise with a digital filter, the noise present in that bandwidth is 10log (40 MHz/500 Hz), or 49 dB lower. This means that the minimum discernible signal should be 77 + 49 dB, or 126 dB below the full scale of the A/D converter. The full-scale of the A/D converter and preamp is about -6 dBm, so the minimum discernible signal is actually -6 minus 126 or -132 dBm, which compares favorably with conventional receivers performing most of the filtering in the analog domain, and using a lower-speed, higher-resolution A/D converter at the end of the signal path.

So what's the downside to this all-digital approach? Unlike the other transceivers mentioned earlier, the Perseus requires a PC and software for operation. There is no headphone or speaker jack - the PC sound card generates the audio for the user. And it does not include a tuning knob, since tuning is done by the PC software. However, it is a great step forward in bringing SDR technology to a consumer price point, and from a technology standpoint, illustrates that SDRs can achieve performance comparable to hybrid analog/digital receiver designs.

About the author:
Doug Grant received his first ham radio license from the FCC in 1967 and his BSEE in 1975. He has logged over 30 years in the semiconductor industry, mostly at Analog Devices, where he worked in engineering, marketing, and product line management for a wide range of analog, mixed-signal, RF and wireless products. He has also logged over 500,000 two-way contacts with other radio hams in every country in the world. Doug is currently an independent consultant specializing in semiconductor and wireless technologies.