The "Soft" Radio

Here is a look at how Software Defined Radios will change the way we think about radio communications.

By Gerald Youngblood,* AC5OG

Your editor's first extensive introduction to software-defined radio (SDR) theory was at last year's Central States VHF Society Conference in Ft. Worth, Texas, where AC5OG made a challenging introductory presentation on the subject. I asked Gerald to write a more extensive article for CQ VHF *than the one he published in the conference's* Proceedings, *and he obliged with this material. While this article is illustrative of a 40-meter application, as Gerald points out, the beauty of the SDR radio design is instant reprogramming to the frequency of your choice.*

This piece also provides an overview of Gerald's four-part series which began in the July/August 2002 issue of QEX. *Our thanks and appreciation go to* QEX *Editor Doug Smith, KF6DX, for his cooperation and assistance to Gerald in his preparation of this article.* —*N6CL*

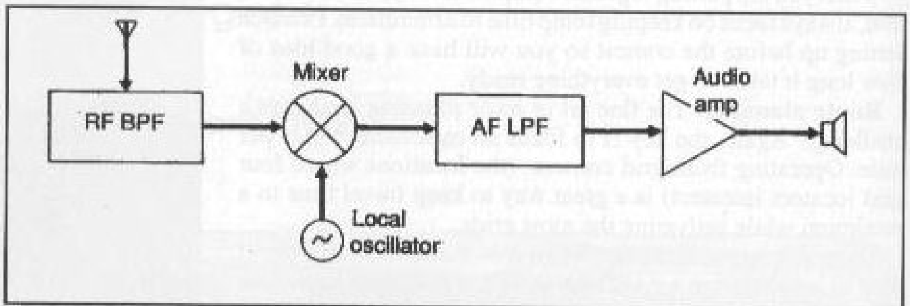

Figure 1. Simple direct-conversion receiver.

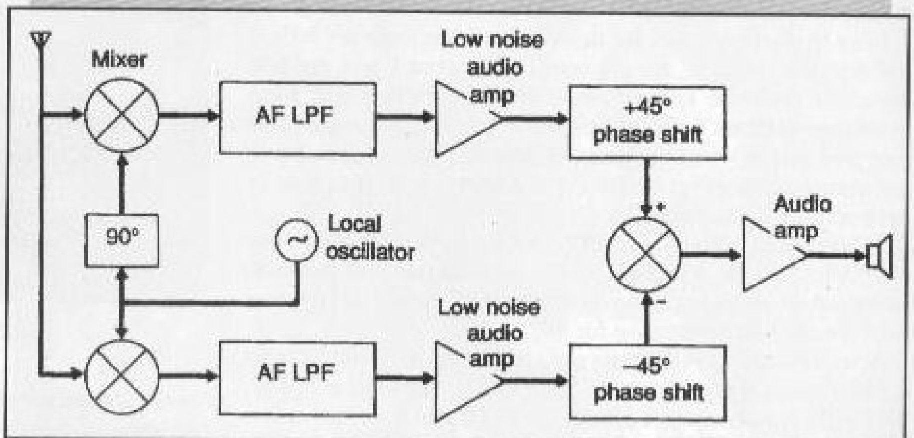

Figure 2. Image-reject receiver.

A revolution is occurring that will change the way amateur and commercial radios are built, used, and maintained. Since the beginning of radio communications, the functionality and performance of a radio was virtually fixed from the time it was manufactured. A design began with a set of specifications that were, in turn, converted into schematics that were subsequently converted to printed-circuit-board assemblies. Aside from hard-wired field modifications, there wasn't much that could be improved or upgraded.

More recently, most amateur and commercial radios incorporate *digital signal processing (DSP)* technology to provide improved filtering and noise reduction. Typically, these have been designed around traditional multi-conversion (superheterodyne) radios, with DSP chips replacing the last intermediate frequency (IF) stage at a low frequency such as 40 kHz. The performance and functionality of the DSP radio is still fixed from the design stage and therefore cannot be modified by the user.

Enter *software-defined radios (SDRs)*. The SDR is a radio in which most or all frequency control, modulation/demodulation formats, and bandwidths are *defined in software* and thus can be changed after the construction process. One should think of SDR hardware as being almost universal in that virtually any radio function may be implemented simply by modifying the software.

In commercial radios this offers tremendous flexibility and cost savings when upgrading to new modes of operation, as is being demonstrated in new wireless-phone base stations. For the amateur radio operator, it means that over time one radio may perform many enhanced functions such as new data, weak-signal, and digital voice modes.

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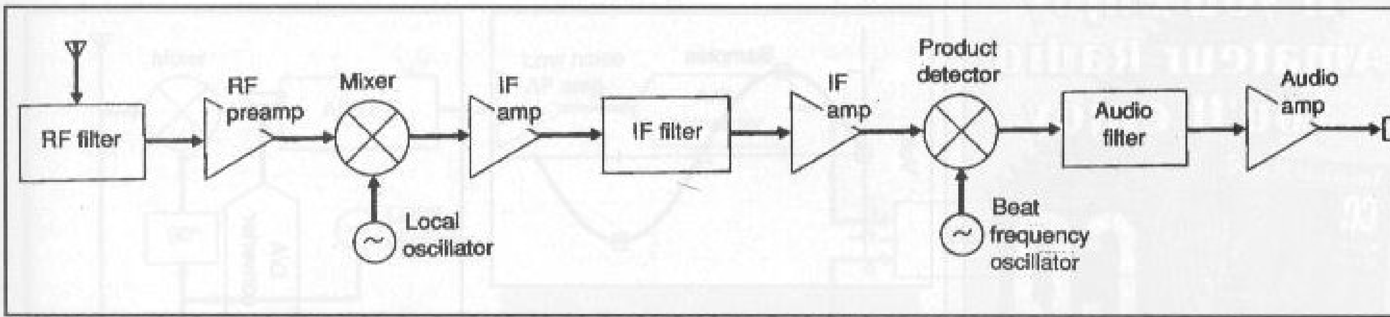


Figure 3. Superheterodyne or filter-method receiver.

SDRs also offer the potential for significant improvement in price/performance over traditional analog radios.

A Look Back at Hardware Radios

To understand SDR concepts, let's first take a look back at the primary hardware architectures that have dominated radio construction. In this article I will focus on receiver technology, because in many cases transmission is the reverse operation of receiving. Virtually all of the principles used in DSP and SDR technologies are analog concepts converted to digital. The principles and the math are the same; however, we can do things with much more precision and flexibility in software.

Figure 1 illustrates one of the simplest receiver architectures—the direct-conversion, or D-C, receiver. The RF carrier from the antenna is bandpass filtered and mixed directly with a local oscillator tuned to the carrier frequency. The mixing process produces outputs at the sum and difference of the carrier and local-oscillator frequencies. The subsequent low-pass filter easily removes the sum frequency, leaving only the difference frequency at zero Hertz. Not surprisingly, this architecture is also frequently called a *zero IF receiver.*

The D-C receiver is wonderfully simple and offers very high dynamic range if properly designed. One problem with this architecture is that when we mix the local oscillator with the carrier, we generate an unwanted image frequency that cannot be filtered out. This image is actually the opposite of sideband. Unless we are listening to an AM signal, this unwanted image will increase the noise level at the receiver output.

To solve the problem, *image-reject* receivers were created as illustrated in figure 2. These are also referred to as *phasing-type, Weaver-method,* and *quadrature down-conversion receivers.* There are a number of different ways to construct an image-reject receiver, but one common analog method is illustrated here. While this method is not widely used today in commercial amateur radio gear, it is the primary architecture used in SDR designs.

The image-reject receiver is similar to the D-C receiver in that the RF carrier is directly converted to zero Hertz; however, instead of one mixer, it incorporates two. While both mixers are driven from a single local oscillator, one mixer is driven directly and the other is delayed by 90°. This means we now have two copies of the incoming carrier that are 90° out of phase (or in *quadrature*) with one another.

We then low-pass filter and amplify the respective signals before shifting the phase plus and minus 45°, respectively. The math actually works out so that we now can add or subtract the shifted signals to receive the lower or upper sideband, respectively.

There is a problem, however, in that slight variations in analog component values result in amplitude and phase differences between the two channels that reduce opposite sideband (i.e., image) suppression. An excellent discussion of how this quadrature down conversion works is available online in an article by Richard Lyons entitled "Quadrature Signals: Complex But Not Complicated." It may be downloaded from the web at <www.dspguru.com/info/tutor/quadsig.htm>.

Until the 1960s, the image-reject receiver was very popular. At that point, high-performance filters became available that allowed inexpensive construction of multiple-conversion *superheterodyne* (or *filter-method*) receivers similar to that shown in figure 3. Superheterodyne receivers have two or more mixing stages with intermediate-frequency (IF) filtering and amplification.

With high-quality crystal or mechanical filters, excellent opposite sideband suppression is possible. The tradeoffs in this approach, however, are increased complexity, images caused by mixing the frequencies from multiple stages, and potentially reduced dynamic range compared to the image-reject method.

In a superheterodyne design, the RF carrier is filtered and usually preamplified before mixing the signal to an intermediate frequency such as 9 MHz or 455 kHz. Many modern receivers use first IF frequencies in the 70 MHz range to spread out the image frequencies so that they are much easier to filter. The signal is then amplified and filtered at the IF frequency before being applied to one or more subsequent mixing and filtering stages.

In figure 3 is shown a dual-conversion example where the second conversion is from the IF frequency to audio or zero Hertz (just as we saw in figure 1, except the carrier begins at the IF frequency). This stage is often called a *product detector,* because it is a multiplication process that creates "products" of the *beat-frequency oscillator* and the IF carrier frequency.

That's enough of the history. It's time to move on to SDR architectures and how they relate to the examples we have just discussed. To learn more about receiver architectures before moving on, please read the appropriate sections of the *ARRL Handbook for Radio Amateurs.*[1]

SDR HardwareArchitectures

The first thing that one needs to understand is how an analog radio-frequency signal can be converted to digital numbers to be processed in a computer and translated back to analog again as sound. This is accomplished through a process called *digi-*

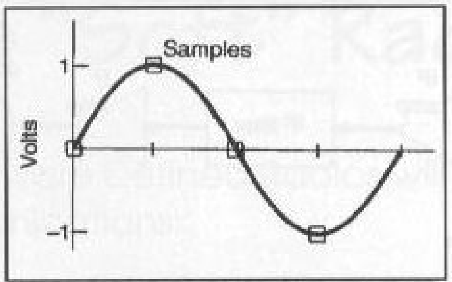


Figure 4. Digital sampling of a sine wave.

tal sampling. (For further study on digital sampling[2, 3], refer to the references provided in the endnotes.)

A radio-frequency carrier can be thought of as a sine-wave voltage that completes a 360° cycle in a time equal to the inverse of the carrier frequency. In 1933 Harry Nyquist proved that one only has to sample a signal at a rate slightly greater than twice the signal's highest frequency component in order to accurately recreate the signal mathematically.

Figure 4 illustrates a sine wave that is sampled at four times the frequency of the waveform. Note that when we sample a 1 V peak sine wave at 0°, 90°, 180°, and 270°, we will measure 0 V, 1 V, 0 V, and −1 V, respectively. If, for example, the sine wave had a frequency of 40 kHz and we sampled the signal at 160 kHz, we would get one sample every 6.25 ms (i.e., 1/160,000).

All SDRs use *analog-to-digital converters,* also called *ADCs,* to digitally sample analog signals so that a computer can process them. To convert back to analog, a *digital-to-analog converter (DAC)* is used. Most computers today include a sound card that allows audio frequencies up to 20 kHz in bandwidth to be recorded and played back by the computer. The sound card digitizes the incoming sound with an ADC and then stores the sampled signal in computer memory.

Applications may then use the samples to perform various types of signal processing such as graphic equalization (tone control) or amateur radio digital modes such as PSK31. In fact, a PC with a reasonable-quality sound card provides much of the hardware needed for a powerful SDR.

SDRs use three primary hardware architectures to convert the radio frequency (RF) signal to analog: *superheterodyne with IF sampling, quadrature*

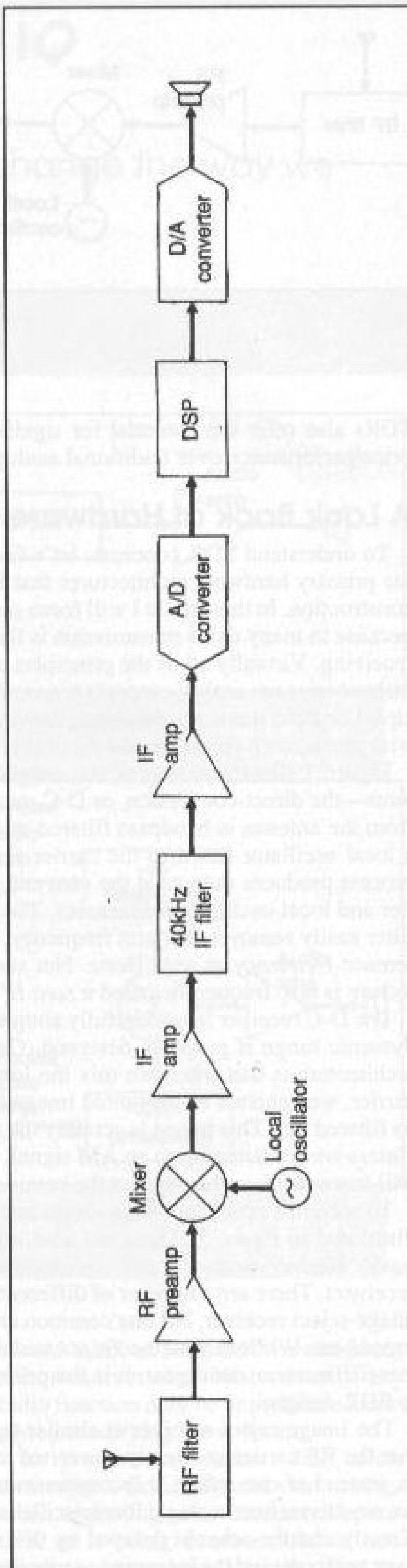


Figure 5. Superheterodyne receiver with low IF sampling.

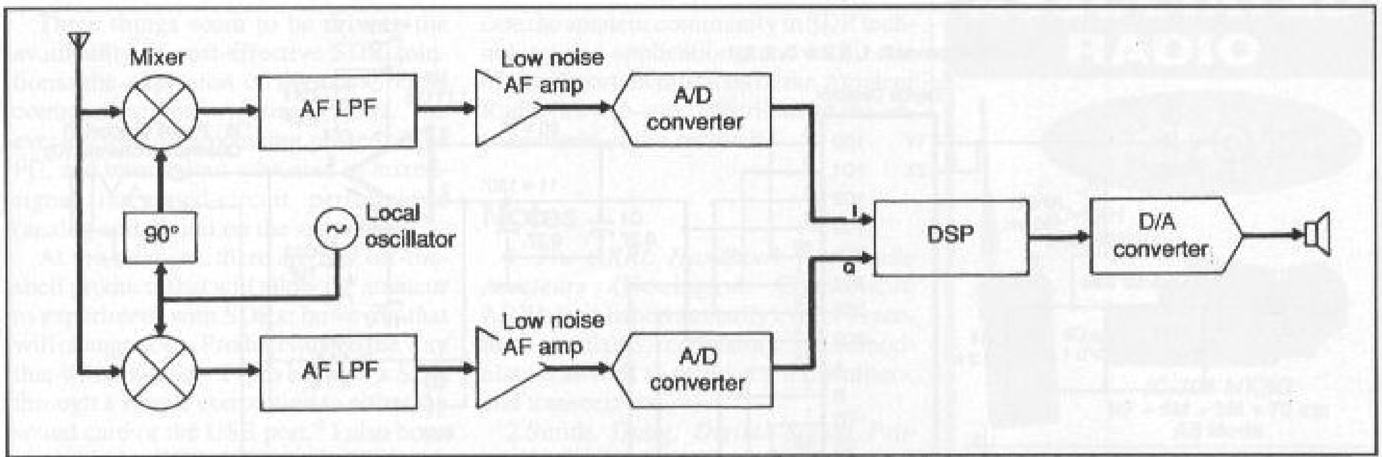


Figure 6. Quadrature down-conversion and sampling receiver.

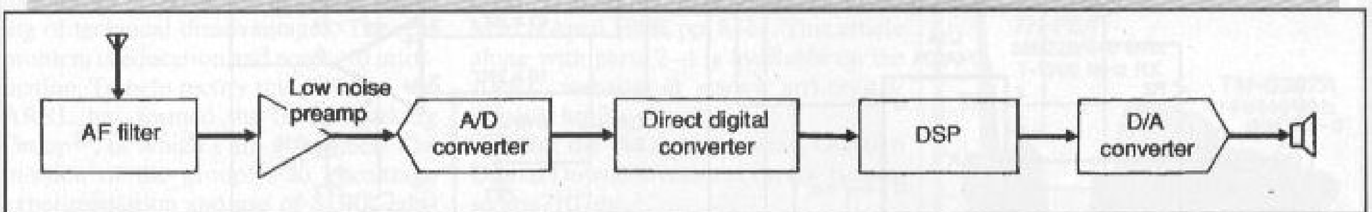


Figure 7. Direct-digital-conversion receiver.

down-conversion to baseband (another word for zero IF as described above), and *direct digital down-conversion (DDC)*. As technology improves, the goal is to move the A/D conversion as close to the antenna as possible so as to remove the imprecise nature of analog circuitry.

The technology now exists to simultaneously sample the entire HF spectrum in real time. The main problem is the dynamic-range limitations of the ADC.

Most DSP transceivers on the market use the superheterodyne with IF sampling architecture similar to that shown in figure 5. The product detector, beat-frequency oscillator, and audio filtering shown in figure 3 are replaced by digital circuitry and software in the digital signal processor. A typical IF frequency might be 40 kHz. After A/D conversion, the signal is mathematically translated from 40 kHz to 0 Hz and filtered using high-performance, flexible digital filters. However, this is really still an analog radio with a digital back end.

The technology really starts to get interesting when we examine the other two methods of conversion. Figure 6 shows a *quadrature down-conversion and sampling receiver* where the A/D conversion replaces the phasing and summation functions shown in figure 2. With this architecture, we are able to convert from RF to quadrature baseband in a single step. With the information from the two channels of digitized signals, we can extract and filter the signals within a bandwidth of two times the sampling rate of the ADCs. *(Note: We are able to double the bandwidth because the two channels effectively double the sampling rate.)*

With the advent of very-high-speed ADCs and DDCs, it is now possible to sample signals at rates of up to at least 100 MHz[4]! This means that with a system similar to the one shown in figure 7, we can simultaneously digitize large chunks of the HF spectrum. We can then digitally down-convert the signal to baseband for subsequent filtering and signal processing.

From the diagram, one can see that virtually every feature of the radio may be controlled and upgraded through software. Such a radio would not become obsolete quickly because new features and enhanced performance are just a software download away.

A PC SDR Example

I first became interested in software-defined radios after reading Doug Smith's 1998 article series in *QEX* (see endnote 3). Noting how PCs with sound cards were being used to deliver new digital modes such as PSK31, I decided to explore how a complete SDR could be built using a PC and a minimal amount of external hardware. I also decided to use Microsoft Visual Basic for the development language to allow rapid development and ease of maintenance.

For this project I chose the quadrature down-conversion and sampling architecture shown in figure 5. I used a very interesting approach called a Tayloe Detector[5] to achieve the quadrature down-conversion. Figure 8 shows the single balanced version of the Tayloe Detector that I used in my first prototype.

The Tayloe Detector has a number of advantages over traditional mixers: In addition to down-conversion, it functions as a very high-Q tracking filter centered at the carrier frequency. It also exhibits an ultra-low noise figure of ~1 dB. I describe the Tayloe Detector as well as some of the basics of digital sig-

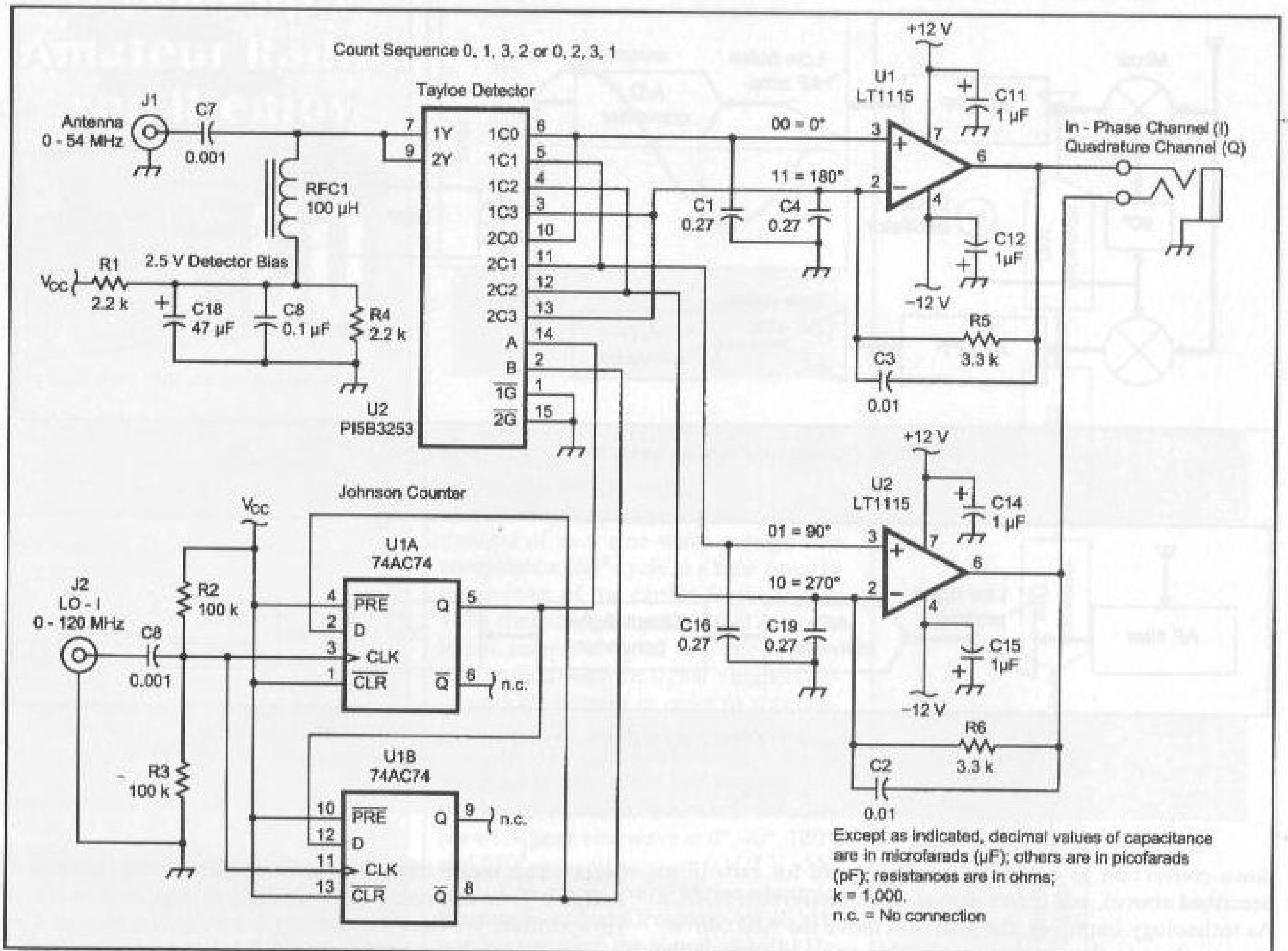


Figure 8. Tayloe Detector quadrature down-conversion. (Artwork courtesy QEX*)*

nal processing in more technical detail in the July/August 2002 issue of *QEX*.[6]

The SDR-1000 transceiver is the result of many hours of study and development. Figure 9 is a screen shot of the front panel showing a 20-kHz wide spectrum display of the 40-meter CW band. A number of CW signals are visible on the display. The user can tune to any one of the signals simply by clicking the mouse on the display. Ten predefined filters are provided, along with separate continuously variable filters for CW and SSB.

Figure 10 is a screen shot that shows the filter shape of the 500-Hz bandpass filter. To generate the display, I fed an antenna noise bridge into the SDR-1000 to fill the spectrum with wideband noise. The SDR-1000 uses a very-high-performance filtering method called FFT[7] fast convolution filtering with 2048 digital filter taps.

As one can see from the display, this is truly a "brick wall" filter. The shape factor from 3 dB to 60 dB is 1.05[8]! Also, note that stop-band attenuation is greater than 120 dB. As a reference, the Inrad 705 crystal filter marketed for the Yaesu FT-1000MP has a 6 dB to 60 dB shape factor of 2.6.

The beauty of the SDR is that there is a virtually endless supply of features and enhancements to be added to the radio. Therein lies the real fun. Many times when I have a new idea, within five minutes to five hours it has been incorporated as a new feature in my radio. Try that with hardware!

Imagine some of the amateur radio applications for SDRs:

- Multimode voice and digital transceiver
- Competition-grade HF-6m radio
- Total digital remote control over the internet
- High-performance IF for microwave use
- Ultra-weak-signal work (below the noise)
- Digital voice modes
- Dream it and code it modifications

SDRs Now and in the Future

SDRs today offer performance and functionality that were heretofore completely out of reach for the average amateur operator or experimenter. Virtually all of the concepts employed in SDRs have been around for many years. Why then is now the time?

Three things seem to be driving the availability of cost-effective SDR solutions: the explosion of digital wireless communications systems (PCS), the ever-increasing processing power of the PC, and tremendous advances in mixed-signal integrated-circuit performance (analog and digital on the same chip).

At the moment, there are few off-the-shelf products that will allow the amateur to experiment with SDRs; however, that will change soon. Products are on the way that will allow any PC to become a SDR through a simple connection to either the sound card or the USB port.[9] I also hope to make the SDR-1000 available in kit form sometime later this year.

What are the disadvantages of SDR technology? I have had a hard time thinking of technical disadvantages. The real problem is education and access to information. To help rectify this problem, the ARRL has formed the SDR Working Group[10], of which I am a member. The mission of the group is to encourage experimentation and use of SDRs, educate the amateur community in SDR technology and applications, and fulfill one of the important missions of the Amateur Radio Service—to "contribute to the advancement of the radio art."

Notes

1. *The ARRL Handbook For Radio Amateurs* (Newington, Connecticut: ARRL, published annually), refer to sections on mixers, modulators, and demodulators as well as receivers, transmitters, and transceivers.

2.Smith, Doug, *Digital Signal Processing Technology* (Newington, Connecticut: ARRL, 2001), pp. 2-1 through 2-16.

3. Smith, Doug, "Signals, Samples and Stuff: A DSP Tutorial (Part 1)," *QEX*, March/April 1998, pp. 8–11. This article along with parts 2–4 is available on the ARRL website at <www.arrl.org/tis/info/sdr.html>.

4. See the Texas Instruments GC4016 Digital Downconverter at <www.ti.com/sc/hpa7107u>.

Learning More

The best way to get started in SDRs is to make up one's mind to do so. For me it was just like learning Morse Code. The hardest part was deciding that I had the ability to do it. Learning and applying it was the fun part. The first thing one needs to do is read more about SDRs. Here are some resources I recommend to help in this task:

Free, or Almost Free

• The ARRL website has a number of SDR articles at <www.arrl.org/tis/info/sdr.html>, including those on the DSP-10 by Bob Larkin, W7PUA, and a great *QEX* series on DSP by Doug Smith, KF6DX. The SDR Working Group has committed to expansion of SDR information available on the ARRL site.

• One of the best free resources about DSP on the web can be found at <www.DSPGuide.com>. It not only explains DSP in terms that are reasonably understandable, it also provides example source code in Basic.

• Another good resource is <www.DSPGuru.com>, which provides a number of good articles, including one (recommended earlier in this article) about quadrature signal processing called "Quadrature Signals: Complex, But Not Complicated," by Richard Lyons.

• Check out Bob Larkin, W7PUA's DSP-10 at <http://www.proaxis.com/~boblark/>.

• If one is into ultra-weak-signal work and EME, then Lief Åsbrink, SM5BSZ's website at <http://ham.te.hik.se/homepage/sm5bsz/> is an excellent resource, especially the part about Linrad.

• Follow my four-part technical article series about the SDR-1000 design in *QEX* beginning with the July/August 2002 issue (see endnote 6).

For Purchase and Deeper Study

• Doug Smith's new book from the ARRL entitled *Digital Signal Processing Technology* provides an excellent introduction to DSP from an amateur radio viewpoint (see endnote 2).

• R. G. Lyons has written one of the more readable technical books on DSP called *Understanding Digital Signal Processing*, from Addison-Wesley, 1997 (Reading, Massachusetts).

• Another excellent book that delves deeper into DSP communications theory is *Digital Signal Processing in Communications Systems* by M. E. Frerking and published by Nostrand Reinhold, 1994 (New York, New York).

What Next?

Start reading some of the excellent resources on the web and keep alert to a number of articles that are in the works. If you are interested in connecting with others who are experimenting in this area, feel free to e-mail me at <AC5OG@arrl.org>.

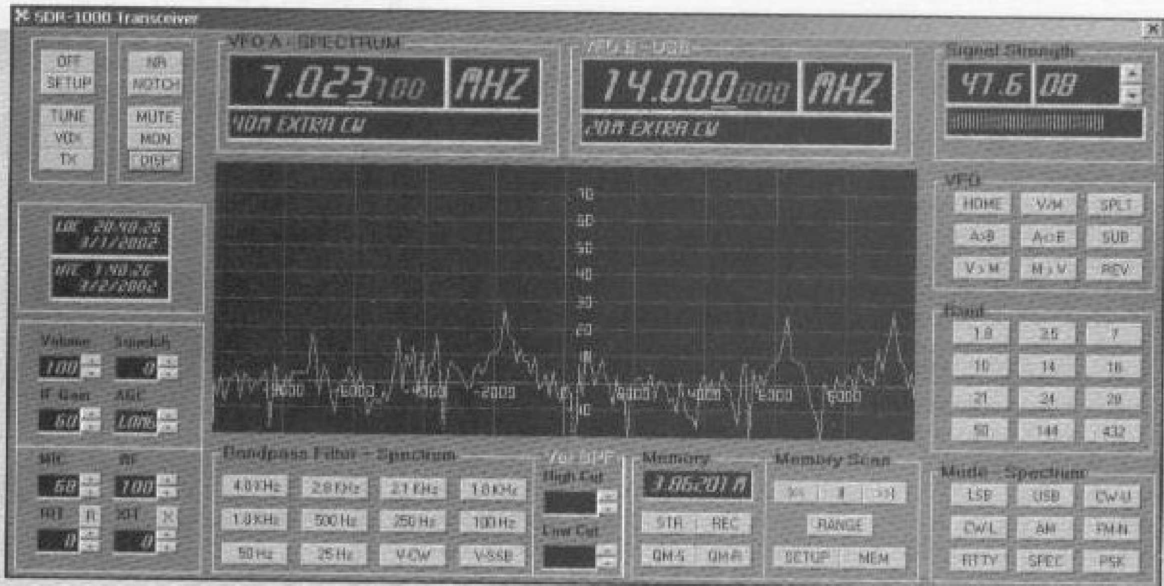
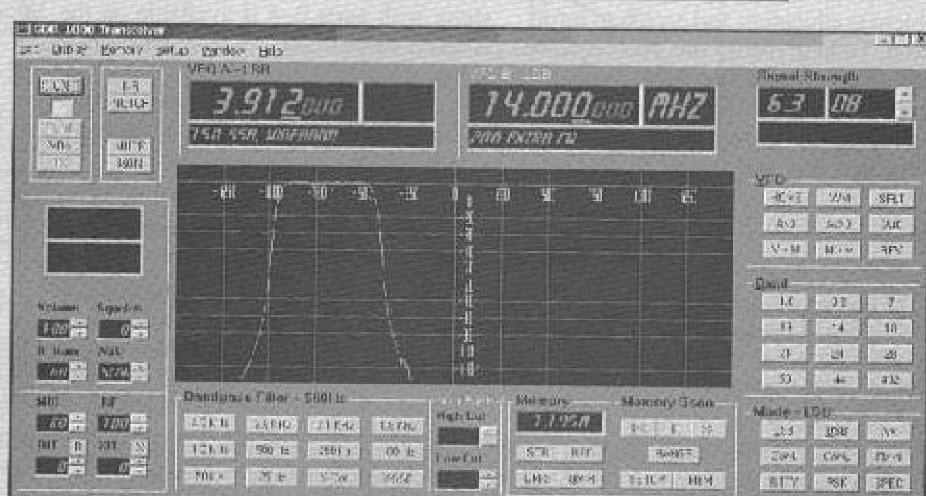


Figure 9. SDR-1000 front-panel interface with wide spectrum display.

Figure 10. 500 Hz filter shape and stop-band attenuation using FFT fast convolution filtering.



5. Tayloe, Dan, N7VE, "Letters to the Editor, Notes on 'Ideal' Commutating Mixers (November/December 1999)," *QEX*, March/April 2001, p. 61. Essentially the same circuit was described earlier in an article by D. H. van Graas, PAØDEN, "The Fourth Method: Generating and Detecting SSB Signals," *QEX*, September 1990, pp. 7–11. This circuit is very close to a Tayloe Detector, but has a lot of unnecessary components.

6. Youngblood, Gerald, AC5OG, "A Software Defined Radio for the Masses, Part 1," *QEX*, July/August 2002.

7. *FFT*, or *fast Fourier transform*, is a method of signal sampling originally developed by French mathematician Baron Jean Baptiste Joseph Fourier in ca. 1807. The equation known as the Fourier transform is a mathematical process in which one can separate any periodic function into underlying sine and cosine functions. In April 1965, by way of attempting to filter noisy signals, Bell Laboratories researchers James W. Cooley and John W. Tukey gathered research that led to the development of the discrete Fourier transform (DFT), which later became known as the fast Fourier transform (FFT), a method of Fourier analysis which decreased the number of calculations needed to analyze the waveform, thus making it infinitely easier to factor out "noise" from the waveforms being analyzed. In this SDR application, FFT works like a spectrum analyzer to convert signals from the time domain (similar to that viewed on an oscilloscope) to the frequency domain. See the website <http://www.netnam.vn/unescocourse/computervision/chap9.doc> for a more in-depth explanation of FFT.

8. Many manufacturers rate their filters from 6 dB to 60 dB. The 3-dB point is actually a better measurement of the noise-power bandwidth of a filter and is a more stringent requirement than the 6-dB point.

9. Expanded Spectrum Systems has just released a kit called The Time Machine which provides a straightforward quadrature mixer that will allow SDR experimentation. It currently runs at fixed crystal frequencies, but input filters are available for most of the amateur bands. More information is available at <http://www.expandedspectrumsystems.com/>. Another interesting system uses the direct digital conversion approach with the AD664 and AD6620 sampling at 65 MHz. It will communicate with the PC through a USB port. The web link is <http://www.rfspace.com>.

10. Members of the ARRL SDR Working Group include Lief Åsbrink, SM5BSZ; Gary Barbour, AC5WO; Bob Larkin, W7PUA; Mike Marcus, N3JMM; Douglas Smith, KF6DX; and Gerald Youngblood, AC5OG. ■