An (almost) all-digital HF communications receiver

CMOS monolithic digital downconverters permit the construction of low-cost receivers using digital signal processing.

By Peter Traneus Anderson

Communications receivers for the high-frequency (HF) band from 3–30 MHz are asked to perform under difficult conditions, extracting weak signals on frequencies close to the frequencies of unwanted strong signals. These receivers use analog circuitry in the signal path, with basic designs descended from Armstrong's original superheterodyne design [1].

Over the past few years, monolithic analog-to-digital converters (ADCs) and digital downconverters (DDCs) have appeared on the commercial market. These devices permit the construction of inexpensive HF receivers using a signal path that is almost entirely digital.

This article considers HF communications receivers used primarily for aural reception of single-sideband (SSB) amplitude-modulated voice signals and on-off keyed (OOK) continuous-wave (CW) Morse-code signals.

Receiver Type	Minimum Discernible Signal (MDS)	Blocking Dynamic Range (BDR)	Intermod Dynamic Range (IMDDR)
Analog	–128 dBm	142 dB	97 dB
ADC	–113 dBm	111 dB	80 dB

Table 1. Analog receiver vs. ADC performance at 14 MHz with 500 Hz bandwidth.

Architecture of digital HF receiver

The block diagram of a basic digital receiver for aural reception is shown in Figure 1 [2][3]. The signal from the antenna passes through a lowpass filter (LPF), a low-gain broadband amplifier and a second LPF to the input of the ADC.

The ADC digital output goes to the input of the DDC. The DDC is set to output a real signal, rather than the usual in-phase/quadrature (I/Q) complex signal. When operating in real mode, the DDC uses Weaver's method of SSB demodulation [4] to shift the desired signal frequency down to the audio frequency range and to bandpass-filter the signal. In Weaver's method, the incoming HF signal is downconverted to a complex baseband lowpass signal and then upconverted to a real bandpass audio signal.

The DDC also decimates the samples, reducing the high-input sample rate to a rate appropriate for the audio-frequency output. The output of the DDC drives one input of a scaling multiplier. The other input of the multiplier is a gain parameter. The gain parameter is adjusted to compensate for varying signal strengths, to keep the audio output at a desired level.

The output of the multiplier drives a digital-to-analog converter (DAC). The DAC's output passes through an LPF to remove aliases, giving the desired analog audio signal. The audio signal is amplified and sent to the loudspeaker.

In the receiver described in [3], the LPFs pass frequencies below 22 MHz and have unity gain in the passband. The preamp has a gain of 10 dB. The ADC is 12-bit, capable of operating as high as 65 MSPS (megasamples per second), with a fullscale input (from 50 ohms through a 2:1 step-up transformer considered as part of the ADC) of –2 dBm. The DDC can operate as high as 52 MSPS.

Both the ADC and the DDC operate at 50 MSPS in this receiver. The 50 MSPS sample rate limits the frequency coverage to frequencies below 22 MHz.

Dynamic range of digital vs. analog receivers

The dynamic range of a digital receiver is limited primarily by the ADC linearity and noise level. Table 1 compares the specified performance of the ADC with the reported measured performance of a high-performance commercial analog receiver, the receiver portion of an amateur-radio transceiver [7].

The transceiver's data in Table 1 comes directly from a product review by Lindquist and Swanson [8]. The ADC data is calculated from specifications in a data sheet [5]. Both receivers are operating at a signal frequency of 14 MHz with a bandwidth of 500 Hz. The transceiver is operating with its RF preamp turned off. The ADC has a 2:1 step-up transformer at its input, and is operating at 50 MSPS.

The blocking dynamic range (BDR) is taken as the signal-to-noise ratio (SNR) in dB at the ADC output of a signal 1 dB down from full-scale, increased by the SNR process gain in dB incurred in the DDC (where the signal bandwidth is reduced from 25

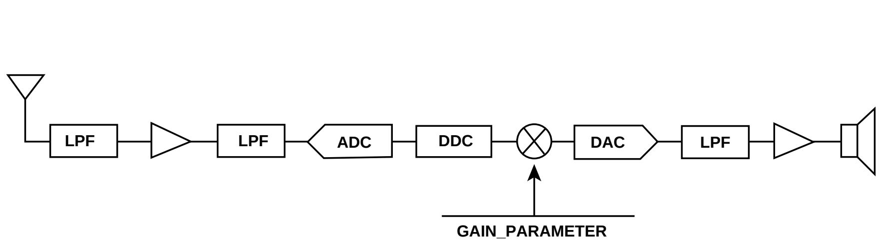

Figure 1. Block diagram of digital receiver for aural reception.

MHz to 500 Hz). The minimum discernible signal (MDS) is taken as the full-scale signal in dBm minus the BDR. The third-order intermodulation-distortion dynamic range (IMDDR) is taken directly from the ADC's data sheet.

The ADC is one of the best currently available, and its performance is clearly not as good as that of the transceiver. The ADC's IMD dynamic range is 17 dB worse than that of the transceiver. According to the ADC's data sheet, dither (out-of passband noise) can be applied to the ADC to improve its IMD performance by 10 dB, giving a spurious-free dynamic range (SFDR) of 90 dB. This is just 7 dB poorer than the performance of the transceiver.

The ADC's MDS can easily be improved in a receiver by providing some gain in a preamplifier ahead of the ADC. The receiver described in [3] uses 10 dB of gain for this reason. The preamplifier can, of course, degrade the intermodulation distortion (IMD) performance of the ADC.

The ADC's BDR is determined by SNR in the ADC's data output. Improving the BDR requires improving the SNR. Improving the SNR requires increasing the number of bits in the ADC data word. This requires the least-significant bit (LSB) becoming smaller, the fullscale signal becoming larger, or both.

The ADC's SNR is 67 dB, good for a 12-bit converter. To equal the transceiver's performance, the ADC's SNR must increase by 31 dB, implying an increase in the word length of 5 bits (at 6 dB per bit). This would give a word length of 17 bits and an SNR of 98 dB.

A longer word implies a smaller LSB or a larger fullscale or both. In a sampling ADC, the LSB cannot be made much smaller than the value in the ADC because of the analog noise floor set by wideband analog noise being aliased into the ADC passband by the sampler in the ADC.

The Johnson noise of a 50 W room-temperature signal source is about –138 dBm for a noise bandwidth of 500 Hz. Thus the ADC's noise floor, –113 dBm for a noise bandwidth of 500 Hz, is only 25 dB above the Johnson noise level.

For the overall receiver to have a noise floor limited by Johnson noise of the 50 W input signal source, there must be some gain and filtering between the receiver input and the ADC analog input, so the bandlimited amplified input noise will be large compared to the ADC internal noise.

Reducing the noise floor by 11 dB might help. This leaves 20 dB to be gained by increasing the fullscale input by a factor of 10. The ADC has a full-scale input at the chip of 1 V peak-to-peak. Increasing the fullscale input to 10 V peak-to-peak is difficult in modern mixed-signal chips, as the trend is to supply rails of 5 V or less.

Another alternative is the oversampling, or deltasigma, ADC. This type of ADC has built-in antialiasing because of the high sample rates used, so the in-band LSB noise can be

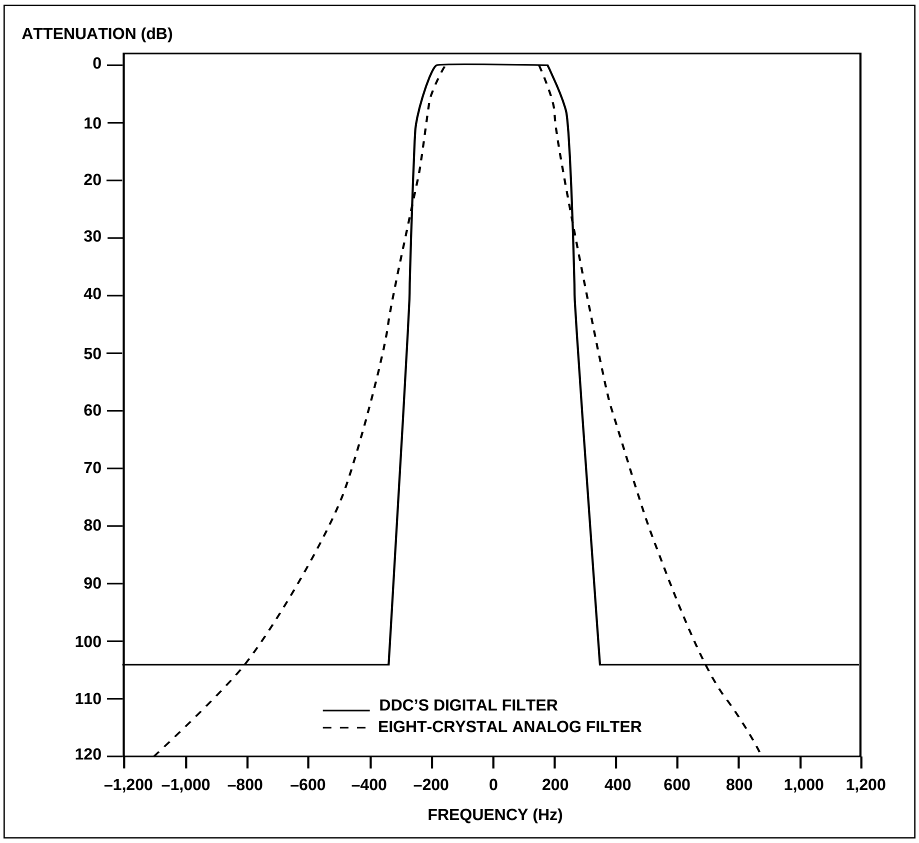

Figure 2. Frequency responses of eight-crystal analog filter (dashed line) and the DDC's digital filter (solid line). The digital filter attenuation floor is actually ragged, always below the straight lines shown.

pushed down to the Johnson-noise floor. Deltasigma ADCs are well-known for potential high SNR and linearity.

Adams, et. al., describes an audio deltasigma ADC with 100 dB of SNR available at an oversampling ratio of 64 [9][10]. Recent experiments have demonstrated the potential of deltasigma ADCs at HF signal frequencies. Gao, et. al., describes a registered comparator intended for use in deltasigma ADCs at a clock speed of 5 GHz [11]. With an oversampling ratio of 64, the signal frequency range would be DC to 5 GHz/128 = 39 MHz. Jensen, et. al., describes a working deltasigma ADC running with a clock speed of 3.2 GHz and an oversampling ratio of 32 [12]. This gives a signal frequency range of DC to 50 MHz. This ADC has an SFDR of 72 dB and an SNR of 55 dB using a second-order design. Higher orders and higher oversampling ratios will lead to improved performance.

Oscillator phase noise and tuning steps

A superheterodyne analog receiver must have at least one variable-frequency oscillator (VFO) to permit tuning to the desired input frequency. This oscillator is always a difficult part of analog receiver design. Mechanically-tuned VFOs are difficult to control digitally, and are difficult to set to precise frequencies. Phase-locked-loop (PLL) frequency synthesizers have difficult trade-offs between phase noise and tuning step size. Direct digital synthesizers (DDSs) have no problem with small tuning steps, but are limited by the noise and nonlinearities in the DAC used to convert the digital sinewave to analog. The DDC sidesteps all these problems by using a DDS without a DAC.

Characteristics of digital vs. analog filters

Every HF communications receiver exhibits a narrow passband, set by lowpass or bandpass filters. The narrowest filters are used for CW reception. Figure 2 compares the frequency responses of analog and digital CW filters. The dashed line shows the calculated frequency response of an analog bandpass filter using eight quartz crystals, having a –3 dB bandwidth of 400 Hz. The filter consists of two identical four-crystal sections cascaded with an isolating buffer between the sections. Each four-crystal section is a Cohn filter [13][14]. The measured attenuation in dB of one section, was doubled to give the calculated attenuation of two sections cascaded. The solid line shows the specified bandpass response of the DDC operating in real mode. The DDC's filter is set for a –3 dB bandwidth of 427 Hz.

The digital filter shows an attenuation floor at –104 dB not present in the analog filter. The floor is actually ragged, always below the straight lines shown. The floor is caused by the design of the finite impulse response (FIR) filter and to the finite-precision integer arithmetic used to implement the filter. The level of the floor in the digital filter is set by economic tradeoffs (a lower floor requires a more complex chip), rather than by physical feasibility. The digital filter response is wider in the passband and narrower in the stopband, exhibiting the narrow skirts common in digital filters.

Improvements needed in DDC's for HF receivers

Once excellent ADCs, with dynamic ranges as good as or better than those of analog receivers, are available, the DDC performance will have to be improved. The dynamic range must be improved, and the real-mode audio passband must be made shiftable for receiving various signals.

Dynamic range

From Table 1, the analog receiver BDR is at least 142 dB for a receiver bandwidth of 500 Hz. Some applications use much smaller bandwidths. Coherent CW uses time-quantized transmitter keying and synchronous (hence the name coherent) receiver baseband sampling to communicate by Morse code at 12 words per minute using a noise bandwidth of 9 Hz [15]. The narrower bandwidth increases the SNR process gain by 18 dB, giving an analog receiver BDR of 160 dB. A good DDC must have an SFDR greater than the intended receiver BDR, as DDC spurious outputs appear after the process gain in SNR that comes with narrowband filtering in the DDC. Thus, a good DDC for HF radios must have an SFDR of 160 dB or better. The DDC is the best currently available, with an SFDR of 102 dB [6]. On-air listening tests of the receiver in reference [2] showed that this is insufficient: the DDC exhibits a center-of-passband spurious tone 102 dB down from the fullscale signal (note that this is within the DDC's specification limits), which interferes with weak-signal reception.

The spurious tone is caused by nonexact calculations in the DDC. Fortunately, when the DDC's decimation ratio is set to exactly a power of two, the offending tone is absent. With the tone absent, the DDC exhibits output SNR equal to that expected from the ADC's SNR and the bandwidth-reduction SNR process gain. The DDC is useable in an HF receiver because, when used carefully, it exhibits SNR greater than its SFDR.

Fortunately, the restriction on the decimation ratio permits the bandwidths needed for SSB and CW reception. An improved DDC with high SFDR, will permit wider choice of bandwidths, as the DDC's SFDR will be greater than the receiver's SNR at any bandwidth.

Shiftable audio passband

Recall that the DDC down converts the incoming HF signal to a complex baseband lowpass signal, and then

Signal Type	Passband Width	Passband Center	Passband Range
CW	214 Hz	191 Hz	84–298 Hz
CW	427 Hz	382 Hz	168–595 Hz
CW	854 Hz	763 Hz	336–1,190 Hz
CW/SSB	1,709 Hz	1,526 Hz	671–2,380 Hz
NONE	3,418 Hz	3,052 Hz	1,343–4,761 Hz

Table 2. Audio passbands available from DDC at 50 MHz clock.

upconverts the baseband signal to a real bandpass audio signal. In the DDC, the ratio of the center frequency of the real passband to the bandwidth, is fixed. This ratio needs to be made variable, so the center frequency can be varied without changing the passband. Table 2 shows the audio passbands of the receiver in reference [2] for various bandwidths used for SSB voice and CW Morse reception. The 427 Hz and 214 Hz bandwidths should have their passbands moved higher in frequency, so the passband is entirely in the audible range, and so that the passband center is independent of filter bandwidth.

CW passband center frequencies should be adjustable, as different operators prefer to listen to different tone frequencies. Also, the CW center frequency should be independent of bandwidth, so the tone does not change as bandwidth changes. The 1,709 Hz bandwidth, passing frequencies from 671–2,380 Hz, when used for SSB voice reception, is narrower than usually desired. The passband includes barely enough low frequencies and high frequencies for intelligibility [2].

The next wider bandwidth useable with the DDC, is twice 1,709 Hz or 3,418 Hz. This bandwidth doubles the low-frequency band edge to 1,343 Hz. This is too high for voice intelligibility, rendering the wider bandwidth useless. If the passband could be lowered by 1,000 Hz, giving a 344–3,760 Hz passband providing excellent voice reception.

Subjective performance

Because this receiver is intended primarily for aural reception, the subjective response of listeners is significant. Overall, the subjective performance is excellent. Receiver noise levels are below the noise level coming from the antenna. The audio exhibits negligible distortion. The SSB passband, though audibly narrow, is clearly intelligible for both male and female voices. The filter edges are sharp, with no hint of ringing, giving clear response to the on-off keying of CW signals. Mallet [16] reports excellent tuning stability and filter response in listening to CW signals on a similar receiver.

Conclusion

Digital HF receivers are now capable of excellent operation at low cost if the signal environment does not exceed the limited dynamic range and frequency range of the digital receiver. Expected improvements in ADCs and DDCs will soon permit the construction of digital HF receivers with better performance than that obtained in analog receivers. RF

References

1. E. H. Armstrong, "A New System of Short Wave Amplification," *Proceedings of the IRE*, Vol. 9, No. 1, pp. 3-11, February 1921.

2. Peter T. Anderson, "A Simple Synchronous-AM Demodulator and Complete Schematics for the DDC-Based Receiver," *QEX*, pp 3-14, September 1997.

3. Peter T. Anderson, "A Faster and Better ADC for the DDC-Based Receiver," *QEX*, pp 30-32, September/October 1998.

4. D. K. Weaver, "A Third Method of Generation and Detection of Single-Sideband Signals," *Proceedings of the IRE*, Vol. 44, pp. 1703-1705, December 1956.

5. Analog Devices, "AD6640 12-bit, 65 MSPS Monolithic A/D Converter, Preliminary 5/17/97," Analog Devices, Norwood, MA.

6. Harris Semiconductor, "HSP50016 Digital Down Converter, June 1994," Harris Semiconductor, Melbourne, FL.

7. FT-1000MP transceiver, Yaesu USA, Cerritos, CA.

8. R. Lindquist and G. Swanson, "Product Review, Yaesu FT-1000MP MF/HF Transceiver," *QST*, Vol. 80, No. 4, pp. 68-73, April 1996.

9. R. W. Adams, P. F. Ferguson, Jr., A. Ganesan, S. Vincelette, A. Volpe, and R. Libert, "Theory and Practical Implementation of a Fifth-Order Sigma-Delta A/D Converter," *Journal of the Audio Engineering Society*, Vol. 39, No. 7/8, pp. 515-528, July/August 1991.

10. Analog Devices, "AD1878/AD1879 High Performance 16-/18-Bit Sigma-Delta Stereo ADCs," Rev. 0, Analog Devices, Norwood, MA.

11. W. Gao, W. M. Snelgrove, T. Varelas, S. J. Kovacic, and D. L. Harame, "A 5-GHz SiGe HBT Return-to-Zero Comparator," *Proceedings of the 1995 Bipolar/BiCMOS Circuits and Technology Meeting*, pp. 166-169, 1995.

12. J. F. Jensen, G. Raghavan, A. E. Cosand, and R. H. Walden, "A 3.2-GHz Second-Order Delta-Sigma Modulator Implemented in InP HBT Technology," *IEEE Journal of Solid-State Circuits*, Vol. 30, No. 10, pp. 1119-1127, October 1995.

13. S. B. Cohn, "Dissipation Loss in Multiple-Coupled-Resonator Filters," *Proceedings of the IRE*, Vol. 47, No. 8, pp. 1342-1348, August 1959.

14. B. O. Williams, "ARRL Lab Experiments with the Cohn Filter," *QST*, Vol. 71, No. 7, pp. 26-27, July 1987.

15. Headquarters Staff of the American Radio Relay League, "The Radio Amateur's Handbook," *Fifty-Eighth Edition 1981*, The American Radio Relay League, Newington, CT.

16. J.-P. Mallet, "Introduction a la radio numerique, Seconde et derniere partie," *Radio-REF*, France, pp. 19-24, December 1996.

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