Wideband Radios Need Wide Dynamic Range Converters

by Brad Brannon

Wideband receivers typically down-convert from frequencies like 900 MHz to basebands of from 5 to 25 MHz, using a fixed local oscillator—and convert directly to digital. The many individual signal channels within are filtered, demodulated, and processed digitally. Such systems for base stations reduce cost and complexity—they need only a single high-frequency analog front end. But the key link, the A/D converter, must have excellent performance.

A/D specs for wideband receivers are driven by system radio standards. To receive distant signals in the presence of strong nearby signals, a cellular base-station receiver must have wide dynamic range. For example, GSM specs call for receivers that can accurately digitize signals from –13 dBm to –104 dBm in the presence of many other signals (Figure 1)—a 91-dB dynamic range! This implies that the spurious-free dynamic range (SFDR) of the converter and analog front end must be about 95 to 100 dBFS.*

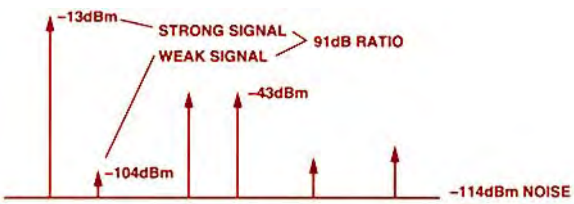

Figure 1. Weak signals must be detectable in the presence of strong signals in nearby GSM channels.

The largest spurs, usually produced by distortion components from strong signals at the front end, could mask weak fringe signals processed by the receiver. The *SFDR* spec permits assessment of signal to noise for signals near the noise floor of the receiver (or SNR's inverse, the bit error rate—BER—in a digital receiver).

GSM is one of the more difficult standards to realize using a broad-band technique, so it serves as an excellent example of the importance of certain converter specifications. Other standards, such as AMPS (North American analog cellular), less demanding on receiver designs, are readily implemented using broad-band.

Full-scale SINAD and SNR, though adequate for single-tone input signals, can't provide the complete picture for the myriad signals and broad bands of spectrum present in wideband radios. Multiple-tone testing and SFDR power sweeps are more informative.

Converters often perform differently when digitizing a full-scale signal than they do with a smaller signal 10, 20, 30 or more dB below full scale—typical of broad-band radios. Figure 2 shows the SFDR of the 12-bit, 50-MSPS AD9042† as a function of signal amplitude. Because of converter integral nonlinearities and track/hold slew-rate limitations at full scale, the SFDR actually improves as the signal level is reduced in the vicinity of full span, providing increased dynamic range. SFDR ratios are better for lower signal levels because the converter is more linear over the rest of the range. Multiple signals also produce near-full-scale codes, but summing of randomized non-correlated signals resembles *dithering.*

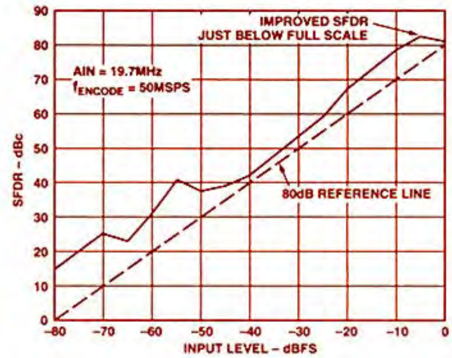

Figure 2. SFDR vs. input amplitude for AD9042.

Dithering is a technique used to lower non-linearities into the effective noise floor by making the converter use different parts of its range each time a given analog level is sampled. It can be implemented by either analog or digital methods. Digitally, a pseudorandom number (*dither*) is generated, converted to analog, and repetitively summed with the analog input signal [so each conversion result for a given level depends on the dither value]. After each conversion, the pseudorandom digital value is subtracted from the digital output. This technique reduces the spectral content that would be generated by repetitively exercising the same nonlinearity. In a wideband receiver, background noise and other non-correlated signals offer some of the benefits of dither, but dither is often intentionally added to improve dynamic performance.

Third Order Intermodulation Distortion: (IMD) is important where there are two large signals in the presence of many smaller signals. The two largest signals will generate spurs caused by non-linearities at $(2f_2 - f_1)$ and $(2f_1 - f_2)$. Significant spurs can override small desired signals located at these frequencies in the same way that harmonics can mask small signals; since these products always fall in band, they cannot be filtered. IMD is not important for its effect on the larger signals, but for interfering with smaller signals in nearby channels. The upper IMD product in Figure 3, aliased back in band, can clearly be seen. Also shown is that, besides IMD, other spurs can present problems. In this case, a large spur at $2(f_2 - f_1)$ indicates that measurements such as two-tone SFDR are just as important as two tone IMD.

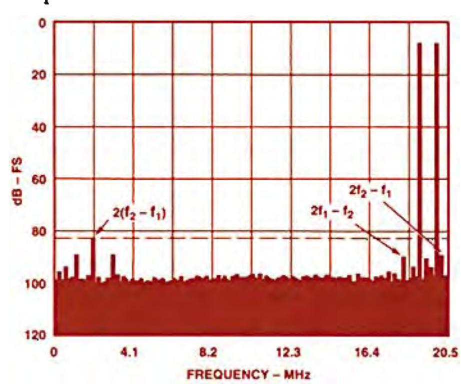

Figure 3. Two-Tone performance of AD9042.

Differential linearity errors (DNL), although architecture-specific, are increased by mismatches within multi-stage converters. They become important when low signal levels straddle a relatively bad code (one that stands out in a DNL plot). The effect can be seen

*SFDR for a converted signal with a given amplitude is the log ratio (dB) of that amplitude to the largest spurious frequency component found in the converter's Nyquist spectrum (0 to $F_S/2$ Hz).

†For technical data on the AD9042 (price \$199 in 1000s – ceramic; plastic available soon), use the reply card. **Circle 3**

in the SFDR plot of Figure 4 by the sharp drop in SFDR between −25 and −40 dBFS. The rms error of the mismatch remains constant, but the SFDR becomes worse as the signal level is reduced and becomes a more-significant contribution to the spurious terms. Further down, the signal no longer crosses these mismatches and the SFDR stays high. Multiple signals or added dither can reduce this error source, improving the receiver's performance.

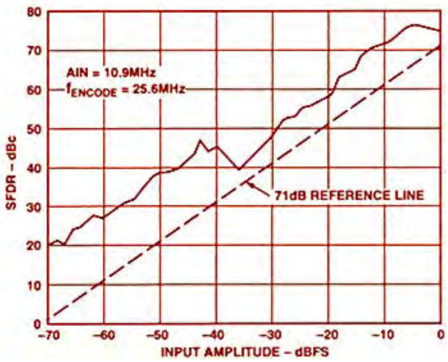

Figure 4. A "bad" SFDR plot. Note decrease near 35 dB.

Head room: When A/D converters receive multiple channels in a broadband architecture, each signal level must be considerably less than full scale of the converter. One signal alone may use the full-scale range of the converter, but when two signals may be present, each must be half-amplitude (−6 dB), assuming equal signal power, to prevent output clipping as these signals sum together at their peaks. *Each doubling of the number of signals requires individual levels to be reduced by 6 dB.* For example, −12 dBFS for 4 channels, −18 dBFS for 8 channels. A multi-channel radio must have enough dynamic range to account for the SNR lost through reduced usable signal levels. In addition, radio designers keep from 3 to 15 dB in reserve as headroom at the top of the ADC range to prevent clipping that comes from inevitable high incoming peak-to-rms ratios and saturation as additional signals come in band as new callers enter the cell zone.

OTHER ADC REQUIREMENTS

Sample rate: Many wide band radios mix down the RF spectrum to baseband (a range of signals from dc to some upper frequency) using wide-dynamic-range, ultra-high-intercept-point mixers such as the AD831 (*Analog Dialogue* 28-2, pp. 3-5). Converters for such radios require a sample rate at least twice the highest frequency (Nyquist rate), i.e., 20 MSPS minimum for signal range from dc to 10 MHz, and generally with at least 20% additional margin, raising the required encode rate to about 25 MSPS.

With both analog and digital standards, oversampling provides a processing gain that improves the effective SNR. For digitally modulated data, the ADC should sample at an integer multiple of the data rate, so that channel centers will fall in the center of FFT or filter bins. For example, if the receiver were decoding GSM packets, the sample rate would be a multiple of the 270.833-kHz data rate. The typical GSM receiver uses a multiple of 48 samples per bit, for a base sample rate, F_S, of 13 MSPS.[1] Sample rates for analog receptions, such as AM and FM, are multiples of the channel bandwidth. With AMPS, a 30-kHz standard, a typical sample rate 1024[2] times higher than the bandwidth is 30.72 MSPS.

Drive and filtering: An alternative to baseband sampling is to sample an IF signal that is in the second or third *Nyquist zone* [i.e., from $(N-1)F_S/2$ to $NF_S/2$]. Thus, the second Nyquist zone is from $F_S/2$ to F_S ; the third is from F_S to $(3/2)F_S$. For F_S = 25 MSPS, the second zone is 12.5 MHz to 25 MHz; the third is 25-37.5 MHz. Using a higher zone can greatly relax the driving amplifier's harmonic requirements because filtering is much easier for frequencies above the first Nyquist zone.

At 10-MHz baseband, for 70-dB harmonic rejection with a 1-MHz signal, the drive amplifier must have 70-dB harmonic performance, because the antialias filter mustn't filter out harmonics below 10 MHz. But if the system were designed for a 1-MHz baseband signal at 26 MHz (F_S + 1 MHz, in the third Nyquist zone) the 2nd harmonic would be at 52 MHz, well outside the 25-to 37.5-MHz passband of the digitizer's anti-alias filter (Figure 5). Converter accuracy need not be sacrificed; all converter harmonics always fall "in-band", due to signal folding within the sampled system. Analog circuit requirements are simplified by the tradeoff of increased amplifier performance for relaxed filter specs. But intermodulation requirements cannot be reduced; IM has to always fall in-band for both amplifiers and converters.

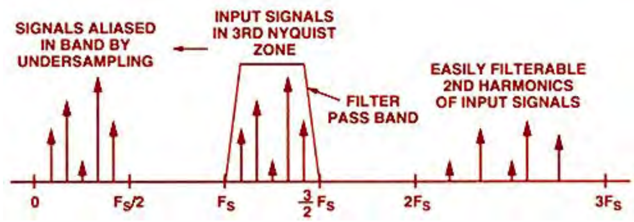

Figure 5. Easy filtering: Undersampling signal in 3rd Nyquist zone to eliminate harmonics at baseband.

[1] Other possible sample frequencies include 26 MSPS and 39 MSPS, both multiples of 13 MSPS.

[2] Other multiples are possible—usually powers of 2 and within the sample rate capabilities of available converters.

WIDE BAND RADIO OVERSAMPLING AND PROCESS GAIN

SNR can be improved by numerical operations called *processing gains*. In any digitizing process, the faster the signal is sampled, the lower the noise floor. The SNR doesn't improve and the total integrated noise remains constant, but it is spread out over more frequencies. The noise floor follows the equation (b = resolution):

$$Noise\ Floor = 6.02\,b + 1.8 + 10\,\log\left(\frac{F_S}{2\,BW}\right)$$

This represents the converter's quantization noise and shows the relationship between noise and the sample rate. Each doubling of the sample rate lowers the effective noise floor 3 dB.

Although some gains are achieved by increasing the sample rate, they are relatively small. However, important gains are achieved in the digital filtering process when it is time to channelize and filter the signals with digital signal-processing chips. For instance, if a 30-kHz AMPS signal is being digitized with an AD9042 sampling at 40.96 MSPS, only a small portion of the broadband noise is passed through the digital filter pass band. The reduction of noise in the pass band, 0.03 MHz/20.48 MHz, is, in log form, 10 log(20.48 MHz/30 kHz), or 28.3 dB.

With this in mind, the effective SNR for a given signal is then

$$SNR = 6.2\,b + 1.8 + 10\,\log\left(\frac{F_S}{(2 \times BW)}\right) - HR$$

If the actual SNR specification is known, substitute it for the (6.02 b +1.8) term. If the converter's SNR spec is 67 dB, with 8 signals, each signal will be 18 + 12 dB (headroom-HR) below full scale (as noted above). Thus, the overall signal levels will be 30 dB below full scale (i.e., SNR reduced to 37 dB). But the *effective* channel SNR will be 67+28.3–30 = 65.3 dB.