DESIGN NOTES

A-to-D Converter Does Frequency Translation – Design Note 259

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The need to characterize frequency sources, both in the laboratory and in the field, is increasingly important. The circuit in Figure 1 offers some interesting attributes in a compact and relatively inexpensive scheme. It uses an LTC®1420 ADC to undersample a higher frequency, driving an LTC1668 DAC, followed by a filter to perform a down conversion. The output of the filter is subsequently resampled to produce a manageable sample rate for a single-chip microcontroller. In addition to characterizing the carrier in an IF strip or the output of a local oscillator, this technique is also useful for characterizing ADCs, DACs, clock sources, signal sources or the effects of logic devices or phase-locked loops on phase noise.

Frequency conversion or translation is usually performed by a diode mixer or a Gilbert cell mixer. Down conversion is most often encountered in radio receivers; up conversion is more commonly used in transmitters. The common superheterodyne receiver usually involves one conversion to produce a fixed intermediate frequency (IF). Spectrum analyzers, cellular base stations, cable modems, microwave and satellite receivers, radar and optical communications systems all include frequency conversion blocks.

Down Conversion with an ADC

It may not be commonly known that down conversion can be performed using an ADC, by undersampling a signal frequency. The resulting output signal frequency is the difference between the sample frequency (f_S) (or a multiple of f_S) and the incoming frequency. An ADC may be used to undersample any frequency that is within its full linear bandwidth.

As in the case of a mixer, the result of this operation is a sum and a difference frequency. The sum frequency, however, ends up at the same apparent frequency as the difference frequency in a discrete time sampled system. Essentially, only the difference frequency remains.

The major constraint in an undersampled system is that the bandwidth of the incoming signals must not fall outside the Nyquist zone in use. (A Nyquist zone extends over a bandwidth of $f_S/2$, above or below an integral multiple of the sample frequency.) Any signal falling outside the desired Nyquist zone wraps back into the DC-to-$f_S/2$ zone. The above constraint can be relaxed if subsequent bandpass filtering in the digital

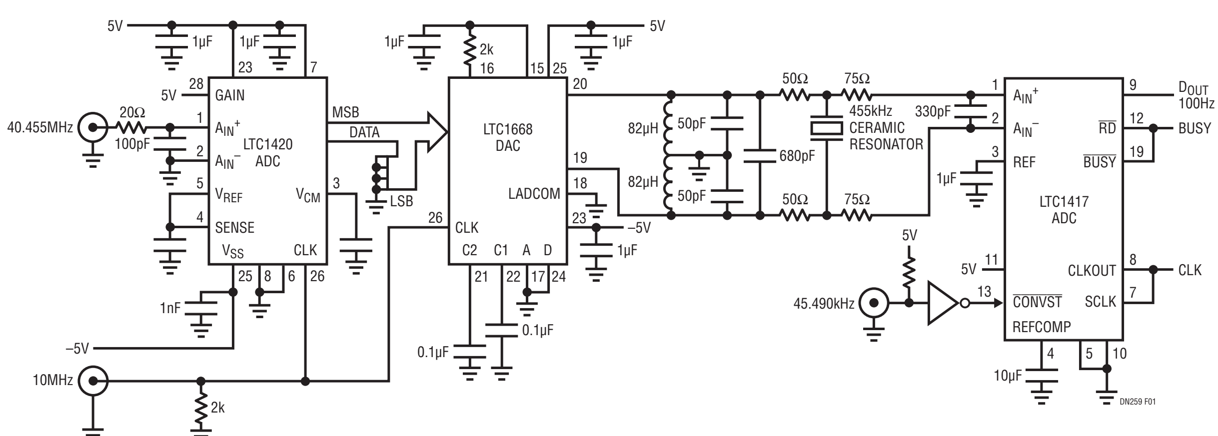

Figure 1. Undersampling 40MHz Performs 2-Stage Frequency Translation to 100Hz

domain limits the frequency range of interest. So long as an unwanted signal does not wrap back into the frequency range of interest, its effect on the spectrum of interest is negligible.

In Figure 1, the 10Msps LTC1420 translates the 40.455MHz input signal to 455kHz at its output. When a high speed DAC is used to reproduce the 455kHz signal, a subsequent analog bandpass filter adds little cost or power dissipation. One advantage of the analog filter is that it does not exhibit mathematical artifacts if the signal frequency is not coherent with the sample rate. In fact, this scheme allows the intermediate frequency to be tailored to suit the conversion rate of the resampling ADC.

An incentive for using a high speed infinite sample-and-hold in this fashion is the benefit of a high sample rate, without the need to process samples at that rate. The data rate delivered by a high speed ADC can be too fast for a low power processor to handle and data rate decimation may reduce signal-to-noise ratio too much. The use of an analog filter after the DAC may seem old fashioned, but the filter characteristics available from ceramic resonators, active filters or tuned LC filters may be hard to match in a digital filter. The use of a higher resolution ADC following the initial 12-bit quantization allows details to be resolved if the original signal contains a few LSB of noise (dither), as well as improving frequency measurement capability.

The 455kHz intermediate signal was chosen to allow the use of readily available 455kHz ceramic resonators or LC filters. Note that the LTC1560-1 monolithic 5th order elliptic lowpass filter could also be used in this application. The LTC1668, as it is a current output device, can drive a tank circuit tuned to the desired frequency. The subsequent resampling of this signal at a submultiple of 455KHz – 100Hz (45,490sps) produces a sinusoid at 100Hz.

In Figure 2, the resampled output of the DAC is shown. Figure 3 is the result of an FFT performed on of the output of the LTC1417.

As mentioned earlier, the output of the DAC is not only the difference frequency of 455kHz. The DAC acts like a mixer and produces in addition to the fundamental (455kHz), the sum and the difference frequencies of the 10MHz conversion clock and the 455kHz signal.

The lower of these unwanted frequencies, 9.545MHz (10MHz – 455kHz), is approximately 20 times the 455kHz or 4.4 octaves above the carrier. The signal level in these components without filtering is approximately 25dB below the carrier; hence, a lowpass or bandpass filter is required. A 2nd order LPF with a 12dB/octave roll-off in the transition region will reduce these unwanted components to approximately 77dB below the carrier, the region of other harmonic and noise components. If the signal under scrutiny is a single tone, a lowpass filter is adequate.

These techniques can also be used on the bench to evaluate the performance of signal generators and clock sources and of course, ADCs and DACs, as well as performing monitoring functions in the field.

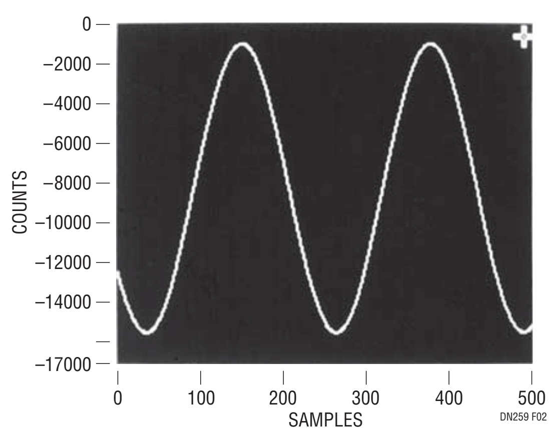

Figure 2. The Downconverted 100Hz Output Exaggerates Phase or Frequency Variation in Original Signal

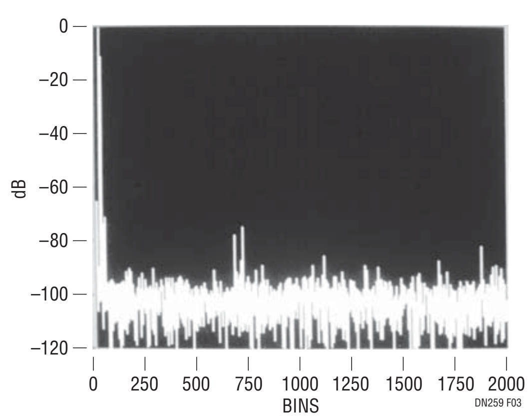

Figure 3. The Spectrum of the 100Hz Signal Can Be Processed to Determine Characteristics of the Original Signal

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