Practical Antialiasing Filters

by Mike Zerkus

To get rid of aliasing in an ADC system, you can use linear active filters, switched-capacitor filters. You also better know when a filter isn't the answer. So, check out Mike's hints for anticipating as well as solving aliasing problems.

Aliasing can contaminate any analog-to-digital process, but it's poorly understood by most engineers. Much of the time, aliasing causes poor data quality. In the worst case, aliased data is accepted and analyzed as fact without anyone ever suspecting that the data is bogus.

But, there's good news, too. Aliasing can be spotted and eliminated if you apply a few simple rules of thumb.

Aliasing is the creation of a false, low-frequency signal. The false signal is the result of an insufficient sample rate of a high-frequency signal.

Aliasing can take many forms. One common example is the apparent backward motion of spoked wagon wheels in old western movies. The wheels appear to spin backward because the camera's frame rate is slower than the wheels' angular velocity.

As the wagon slows down, the wheels apparently change direction, speed up, slow down, and stand still until the wagon is slow enough that the camera can capture the true motion of the wheels. Strictly speaking, this is called temporal aliasing [1].

Spatial aliasing is the jagged edge you sometimes see on lines drawn on a computer screen. A chord played on a musical instrument is another type of aliasing, pleasing to the ear if done correctly.

Of interest here, however, is aliasing that affects A/D conversion. In this article, I assume you have a prior knowledge of ADCs, a general knowledge of filters, and a desire to cut through the theory and get to a few helpful tips. However, if you need a refresher, check out Bob Perrin's "High-Resolution ADCs" (*INK* 74) or *Design of Active Filters with Experiments* [2].

ALIASING IN AN ADC SYSTEM

Aliasing most often appears as a very low-frequency roll in the data, almost indistinguishable from a DC drift. The aliasing is caused by noise whose frequency is greater than the Nyquist frequency.

The noise is undersampled, and the difference between the noise and the Nyquist frequency appears in the data. The amplitude of the false signal depends on the amplitude of the noise.

THE NYQUIST FABLE

The Nyquist rule says that the minimum sample rate required to describe a signal is at least two times the frequency of interest [3]. This rule only works when the signal to be sampled has no frequency component greater than half the sample rate [4].

In the real world, everything has noise. So, every frequency above half the sample rate is aliased or folded back and appears as low-frequency components of the signal. The closer in frequency the noise is to the Nyquist frequency, the lower the frequency of the contamination of the data [5, 6, 7, 8].

Now, suppose you're building an ADC system. The signals you're trying to transduce are between DC and 10 Hz in frequency. Your system is going to operate in a room brightly lit with fluorescent lamps, which are producing 60-Hz noise.

As the sample rate, let's choose 119.95 samples per second. What happens?

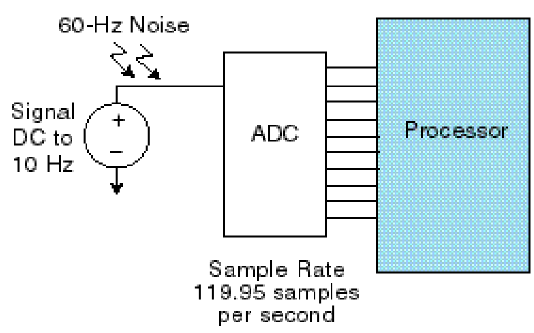

Figure 1—This hypothetical A/D system is set up with a 10-Hz signal and 60-Hz noise

Figure 1 shows a block diagram of the system, and Figure 2 illustrates the noise and signal waveforms. The Nyquist frequency of the system is 59.975 Hz. The 60-Hz noise is undersampled and folds back.

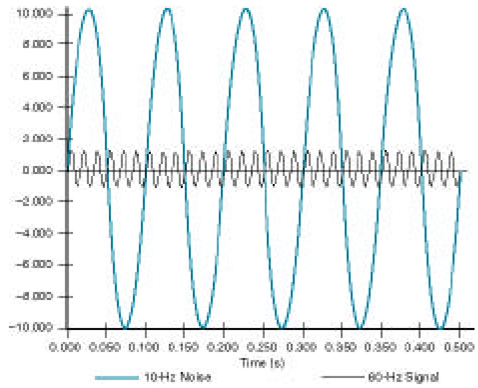

Figure 2—These waveforms (10-Hz input and 60-Hz noise) are input into the A/D system shown in Figure 1.

The beat frequency between the sample rate and the noise (i.e., 60.000 – 59.975 = 0.025 Hz) appears in the data. Because 0.025 Hz is in the frequency range you're transducing, you aren't able to distinguish between the aliased signal and a real signal, as shown by the graphs in Figure 3.

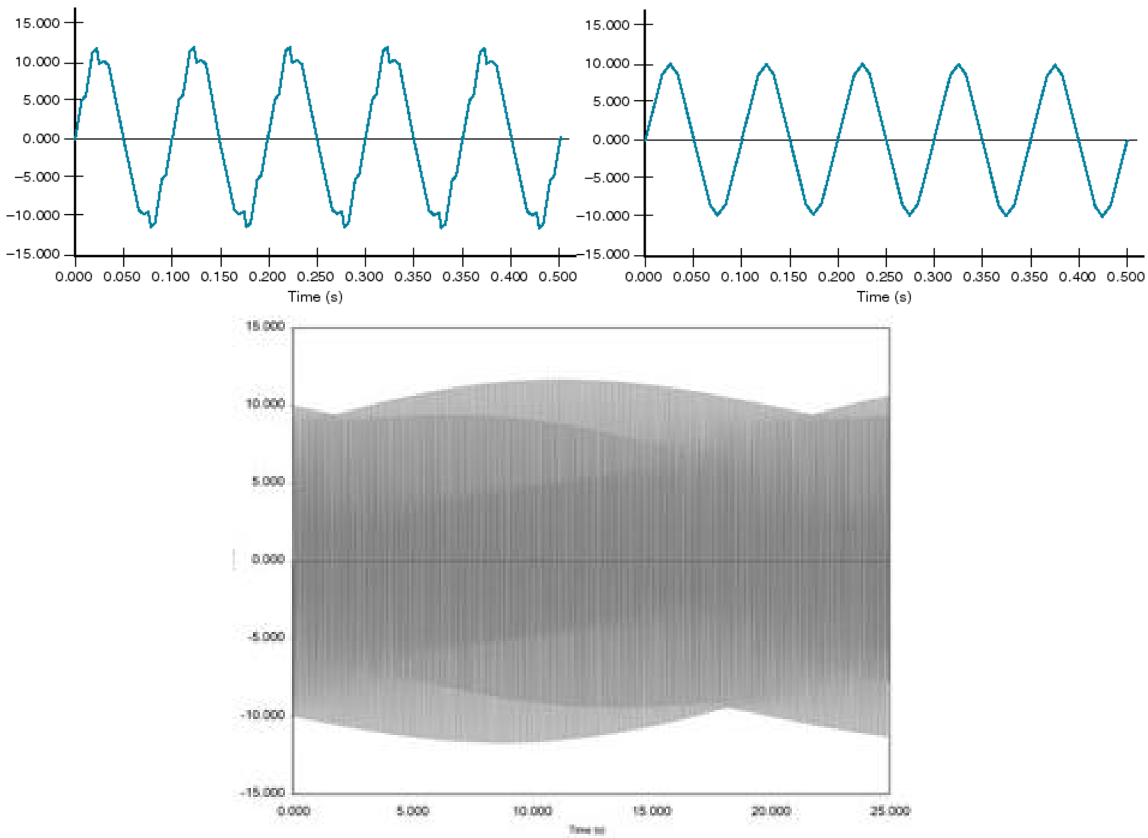

Figure 3—Here's the data as seen by the processor shown in Figure 1. a—The wave actually looks like this, with a 10-Hz signal combined with 60- Hz noise, sampled at 300 samples per second. b—This graph represents 0.5 s of a 10-Hz signal combined with 60-Hz noise, sampled at 119.95 samples per second. Note that there is no apparent distortion of the wave. If we don't know the noise is present, this is exactly what we expect to see. c—This graph represents 25 s of a 10-Hz signal combined with 60-Hz noise, sampled at 119.95 samples per second. The slow roll in the data amplitude, caused by the undersampling of the aliased 60-Hz noise, is only apparent when you view a long data record.

Now, this example may be contrived, but the problem is obvious. In the real world, the noise is never monotonic. The closer the noise frequency is to the Nyquist frequency, the lower the frequency of the aliased contamination.

ALIASING AND SOFTWARE

Contrary to popular belief, it isn't possible to remove aliasing with software. Once the signal is acquired, there's no way to tease apart the real low-frequency components of the signal and any artifacts caused by aliasing [8].

Aliased contamination is sometimes masked by DC offset in the signal as well as genuine low-frequency noise. Keep in mind that, in a system with a sample rate of 10,000 samples per second, 60 Hz is low-frequency noise.

The only real solution to aliasing in a typical ADC system is a good old-fashioned low-pass analog filter [9]. The filter's cut-off frequency should be above the highest frequency to be transduced and below the Nyquist frequency of the system.

There are several approaches to this kind of filter problem. So, let's review them.

PASS BANDS AND POLYNOMIALS

People have written volumes about which filter is the best. What's the bottom line? Go with the flattest pass band and the steepest cutoff at the Nyquist frequency that you can afford.

Of course, as with all things, there are tradeoffs. In most situations, a flat-pass band is more important than a steep cutoff. But, that's only true provided that there is enough cutoff to stop the aliasing.

Figure 4 shows pass- and stop-band characteristics of various filter types [10]. The corner frequencies of the antialiasing filter must be between the highest frequency of interest and the Nyquist frequency.

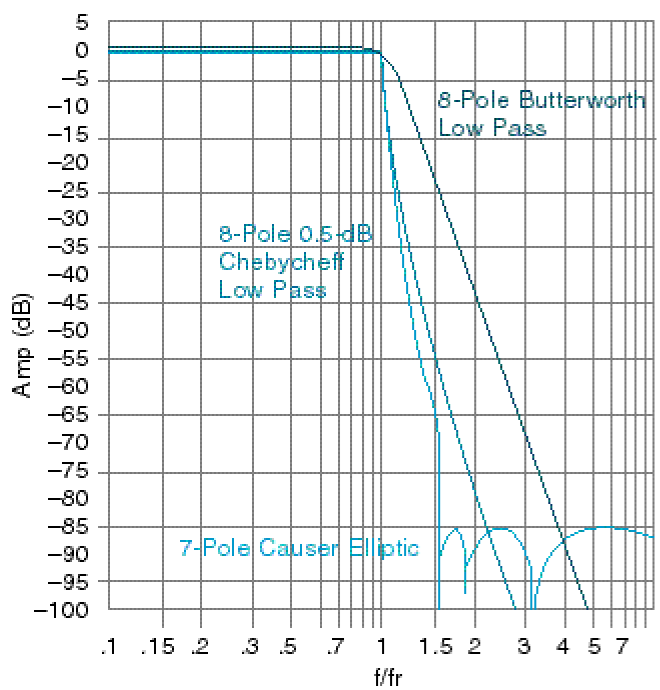

Figure 4—This graph compares typical low-pass filter transfer functions [10].

You want to place the antialiasing filter as close as possible to the input of the ADC. Also, the filter should have as many poles as possible, with eight being a general, practical limit for most systems. The aliased noise must be below the noise limit of the system at the Nyquist frequency.

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If you observe the 10´ rule [4], where the sample rate is chosen to be ten times the highest frequency of interest, then a Butterworth filter is a good all-around choice. If the highest frequency of interest is closer to the Nyquist frequency, an elliptical filter provides a steeper rolloff. But then, you have to live with ripple in the pass and stop bands.

ANTIALISING FILTERS

The simplest antialiasing filter is a capacitor between the signal and return. Strictly speaking, this one-pole RC filter is not the optimal solution (R is the output impedance of the signal source).

However, I think it's important to mention this option because there have been so many times I've been called in after a system was built and there was no hope of redesigning anything—and, naturally, the customer needed the system up and running right away. Hey, a capacitor is better than nothing.

So, use the reactance formulas, pick the best value, and solder it in. Cheap but effective, a well-placed capacitor can reduce aliasing and improve system performance. Sometimes, that's all you can do.

Passive filters introduce the least amount of extra noise into the signal, but there is insertion loss or attenuation of the signal's DC portion [6]. Passive filters are most appropriate for antialiasing where cut-off frequencies exceed 50 kHz or in situations where power is unavailable.

Active filters provide the best all-around solution because they're easy to implement. A variety of active solutions exist, and the filter can have gain greater than one.

DO IT YOURSELF…

Generally, an ADC system has some type of amplifier sitting between the signal source and the ADC, and this amplifier can sometimes serve as a filter.

There are a lot of implementations of active filters using op-amps [1, 11]. Figure 5 shows a Sallen-Key implementation of a fourth-order Butterworth filter I've used with good results.

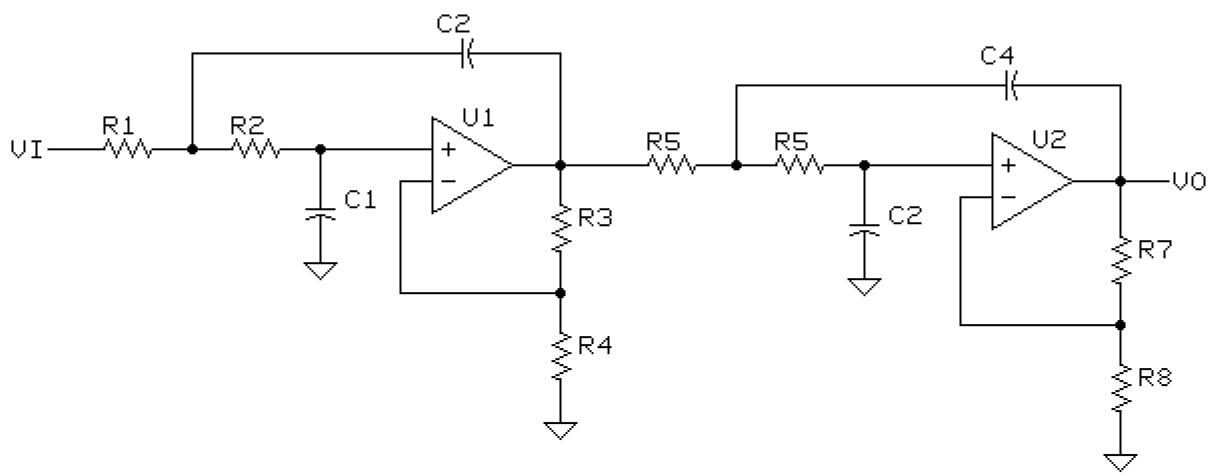

Figure 5—This graph compares typical low-pass filter transfer functions [10].

The trick to getting good results when building up your own active filters is to prototype and test. By building your own filter, you can save a lot in parts, but be sure to weigh this against the added PCB real estate and added design time.

…OR OFF THE SHELF

Some excellent linear active-filter chips are on the market, one good example being the MAX275 from Maxim. As Figure 6 demonstrates, this eight-pole analog active filter can be configured for a variety of transfer functions. Maxim sells a DOS software package to help determine the proper value of passive components.

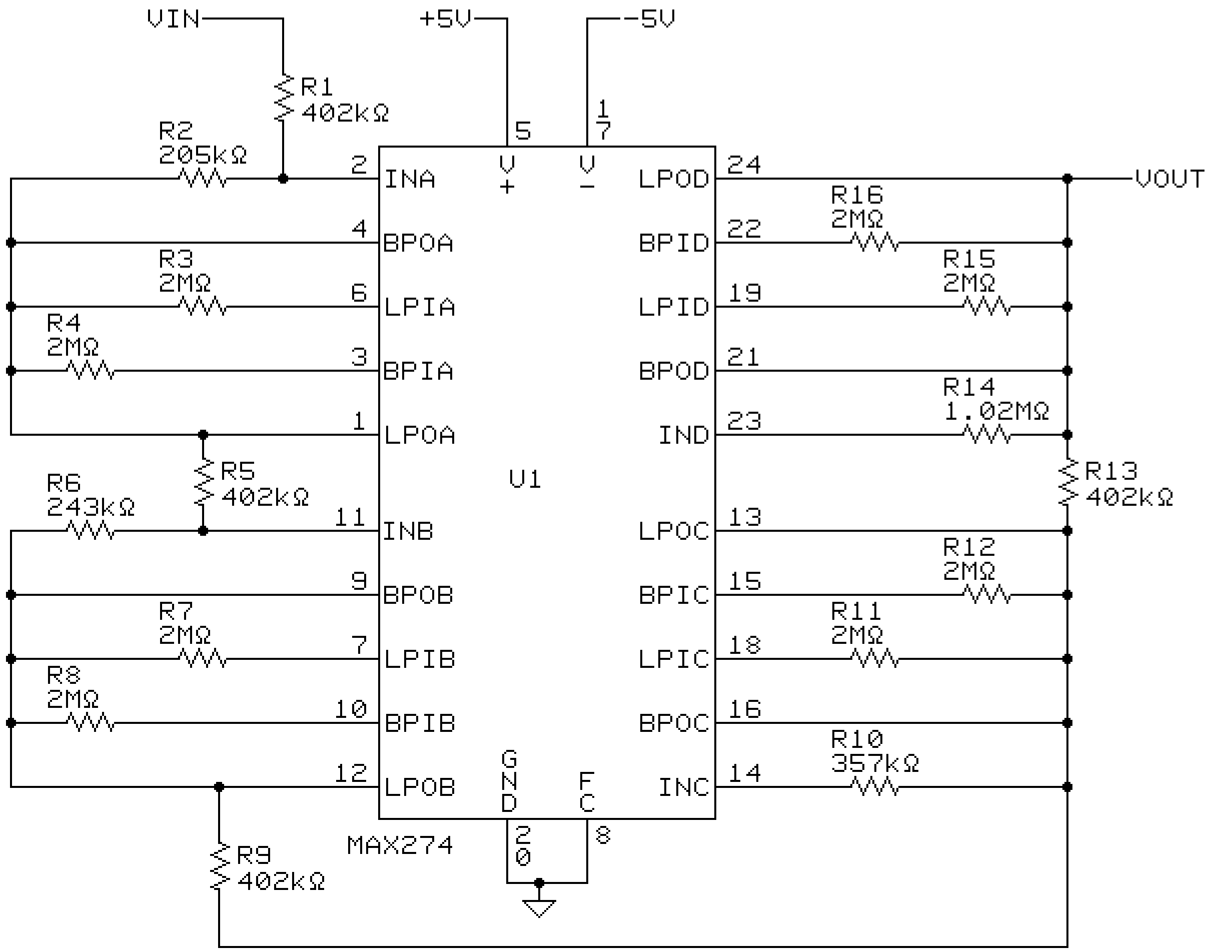

Figure 6—This eight-pole linear active filter (f0=1000 Hz) was built with the MAX275.

Using a linear filter chip can save design time and provide more consistent results. Another company with a great line of filters, Linear Technology, offers a Windows filter design program on CD-ROM.

SWITCHED-CAPACITOR FILTERS

Another option—the switched-capacitor filter—is an active filter that uses electronic switching of a capacitor to imitate a high-order filter. Several manufacturers produce switched-capacitor filter chips.

The filter's cut-off frequency is controlled by a clock frequency applied to the chip, which controls the electronic switch. Typically, the clock frequency is 50–100 times the cutoff frequency of the filter [6, 11].

The major advantage of this setup is that the cutoff frequency is easy to change. So, if the system requires a variable sample rate, the antialiasing filter can follow along.

The major disadvantage is that a switched-capacitor filter is also subject to aliasing [6, 8, 11]. Eliminating the aliasing in the switched-capacitor filter sometimes requires a prefilter in front of the switched-capacitor filter and a reconstruction filter behind it. Some switched-capacitor filter chips have an op-amp linear filter onboard the chip to provide pre- or post-filtering.

Clock feedthrough, which is an extraneous signal that switched-capacitor filters create, can occur in the signal. This feedthrough resides at 50–100 times the filter's corner frequency. The presence of clock feedthrough can cause additional aliasing problems.

Switched-capacitor filter designs work best if the available space is small, there is ample time to debug the design, and especially if the cutoff frequency has to be variable.

"Reducing Noise in a Switched-Capacitor Low-Pass Filter" is an excellent article detailing methods to better apply switched-capacitor filters [12]. Additionally, *The Art of Electronics* includes a detailed example system with a variable cut-off switched-capacitor antialiasing filter [11].

FILTER MODULES>

A filter module is a prebuilt filter unit that doesn’t require any external components or adjustments. Filter modules are expensive, but the results are almost always good.

Two examples of filter modules are the D74 series and the 858 series from Frequency Devices. These are shown in Photo 1.

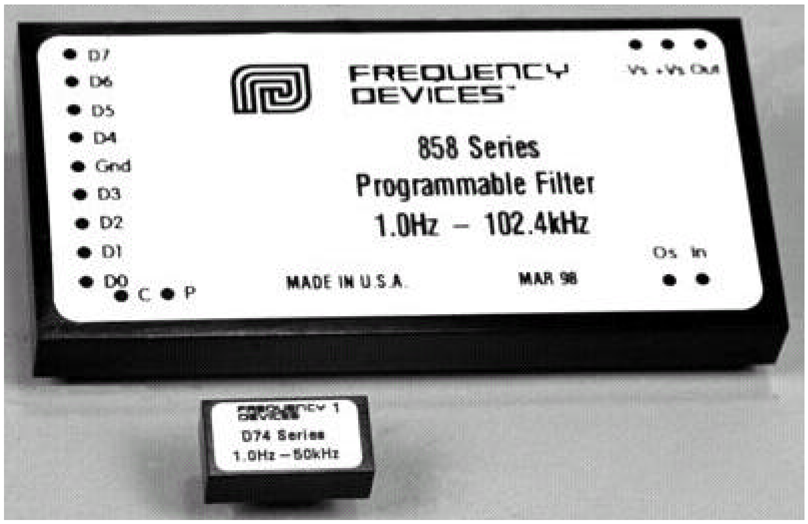

Photo 1—These D74 and 858 series filter modules are manufactured by Frequency Devices

The D74 series are 16-pin DIP modules with a fixed frequency. The 858 series are programmable eight-pole active filters in a plastic module. These devices are available in several response curve formats and frequency ranges. Filter modules represent the easiest, most direct way out of an aliasing problem, but they certainly aren’t the cheapest option.

READING THE VOLTAGE

If the system in question just needs to read a voltage from time to time, your frequency of interest is 0 Hz (DC). This means that your sample rate should be at least 2 ´ 0 Hz = 0 Hz, which doesn’t make sense until you realize that all frequencies are aliased in this situation.

Luckily, this is the one situation where software can help because anything that is not DC (i.e., anything that is moving) is bad.

An analog filter is still a good idea. But in addition to the filter, the ADC can be read several times in a row and all the results are then averaged. The number of samples to average depends on how much time is available, but a good rule of thumb is 100 samples [8].

MULTIPLEXED INPUTS

Some systems have a multiplexed input to a single ADC. In fact, most off-the-shelf ADC boards fall into this category.

These systems come in two types—internal (including microcontrollers with multiple A/D inputs) and external. Figure 7 shows a multiplexed system with an antialiasing filter.

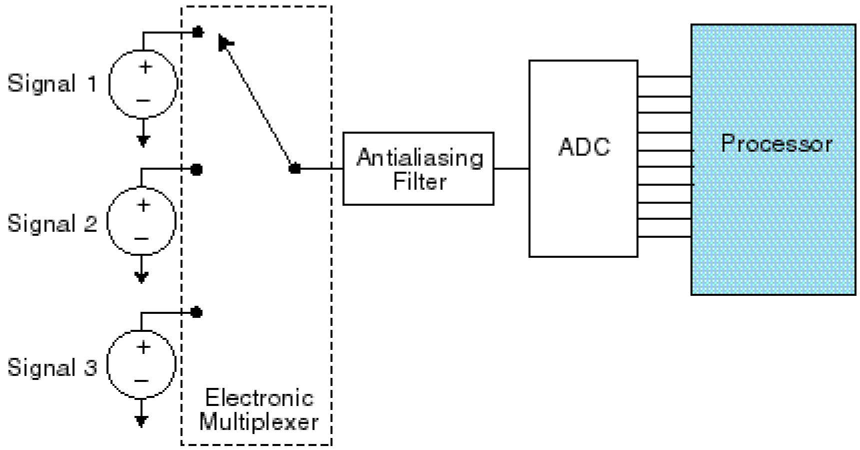

Figure 7—Antialiasing filters can also be used with multiplexed A/D systems.

With an internal multiplexer, the point between the multiplexer's output and the input to the ADC is not accessible. Therefore, the only choice is to put an antialiasing filter on each channel.

However, in a system where the point between the output of the multiplexer and the input to the ADC is accessible, placement of the filter is not so clear. A filter between the multiplexer and the input to the ADC is best if all the input signals are similar.

In a multiplexed system, the signal is sampled twice—once by the multiplexer and once by the ADC.

So, if you have an eight-channel multiplexer scanning eight signals, then by definition, the sample rate of the ADC must be at least eight times higher than the switching rate of the multiplexer. Therefore, the settling time of your filter or amplifier must allow you to read the true signal [8].

ALIASING ALIBI

Sometimes, however, aliasing isn't even the problem. A mechanical imperfection in the system can also produce alias-like effects. Unfortunately, a filter can't help you here.

Instead, consider a tachometer speed sensor with eccentric gears. A DC tachometer is mechanically connected to a shaft by a set of gears (see Figure 8). If the gears are not aligned with the center of the shaft, they will lope (i.e., speed up and slow down) as the shaft spins.

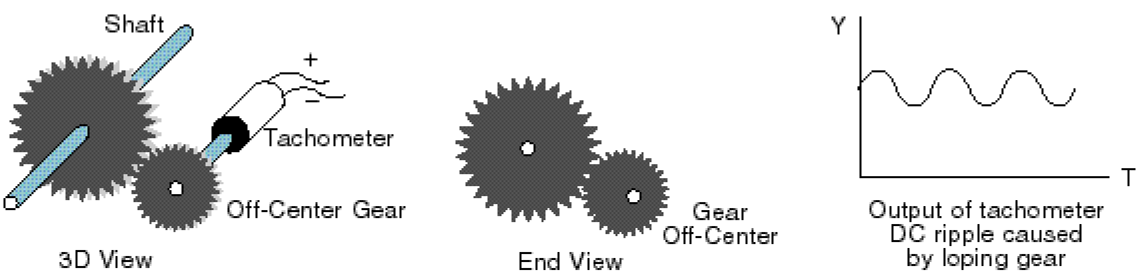

Figure 8—This DC tachometer is conected mechanically to a shaft by a set of gears. The gear on the shaft of the tachometer is eccentric. b—This view of the tachometer gear train shows the off center tachometer gear. c—The output wavefprm with ripple is caused by the gear eccentricity. If the gears are large enough, the ripple could be mistaken for aliasing.

The average velocity of the system is true, but the instantaneous acceleration—and thus the instantaneous velocity—is not constant. The output of the tachometer is a DC voltage with a small AC waveform on top of it, which results from the eccentricity of the gears.

This isn't aliasing in the classic sense. It's the tachometer reporting the truth. I'll spare you a diatribe on my disdain for gears and just point out that the signal looks like it may have alias contamination, but in fact it does not.

Be on the lookout for such situations, especially where the sensors aren't directly coupled to whatever they're monitoring.

RULES OF THUMB

With these rules of thumb, you should be able to focus your efforts whether you're designing a new system or troubleshooting an old one:

- If it is a sampled data system, aliasing can happen. Try not to hang around people who think that aliasing is something that only happens to Sigourney Weaver.

- When in doubt, filter it out. If you're designing a new system, plan on placing an analog low-pass filter as close as possible to the input of the ADC. The filter should have as many poles as you can afford, with eight poles being a general, practical limit for most systems.

- Cut off frequency. The filter's cut-off frequency should be below the Nyquist frequency and above the highest frequency of the input signal.

- Many poles are good, but a cap to ground is better than nothing. Imagine that you're called in to "fix" a problem in an existing system and aliasing is the cause, but for some reason, you can't install a proper analog filter.

- A properly chosen capacitor placed between the ADC input line and signal return provides a one-pole low-pass filter. This solution isn't the best, but it's usually easy to install, it reduces contamination, and you'll come out the hero.

- There is no antialiasing filtering in software—unless you have worked out the software implementation for the Houdini-Copperfield Mystery filter. Once the aliased low-frequency signal gets mixed with the real low-frequency signal, separation is impossible.

- Don't invite trouble. If possible, choose your sampling frequency to be something odd. Avoid choosing the frequency of the predominate noise source or its harmonics for the ADC sample rate.

- In other words, avoid 60 Hz and harmonics of 60 Hz. (Believe me, I wouldn't include this rule here if I hadn't seen it done—more than once.)

- Know your noise. Use all the tricks—shielding, grounding, guard ring, and so on—you have at your disposal to keep the noise out.

- Understand system needs. Overkill is expensive. A design is a series of tradeoffs between conflicting goals.

- Look at the big picture. Aliasing most often appears as a slow DC roll. By looking at a long data record, you can see the periodic nature of the shift.

- Beware the finger-pointers. Make sure there isn't a mechanical or other defect that's producing an artifact that resembles aliasing.

Sure, aliasing can contaminate the data collected from an A/D system. But by applying a few rules, you can anticipate where and when aliasing problems may occur, and you'll have a good idea of the solution as well.

In the words of a wise friend of mine, *per ardua ad astra* (through difficulty to the stars).

Thanks to Kathy Haynes for help with this manuscript. Most of all, I want to thank my father, J.M. Zerkus, for explaining the "wagon wheel" illusion to me.

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