Noise and Vibration Measurement System Basics

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Hardware and software for accurate measurement of noise and vibration signals have evolved considerably over the last decade or so. Until the mid 1990s, multichannel data acquisition systems were relatively expensive. Driven by the audio industry, new electronics have been developed that revolutionize the price/performance of noise and vibration data acquisition systems. Hard disk space has dropped in price so that it is now possible to record and store time histories at a reasonable cost in most cases. In this article, we will concentrate on recommendations to correctly choose and operate a modern dynamic signal measurement system.

Signal Conditioning. A simple data acquisition channel is shown in Figure 1. The schematic shows only a single channel, and it supports our discussion of signal flow through some important steps. Electrical details such as single-ended or differential input coupling are omitted in this discussion. Also not shown is the signal conditioning that is usually located prior to AC/DC coupling, such as IEPE current supply for sensors with built-in impedance conversion.

The AC/DC coupling is used to remove any constant voltage (DC) from the sensor or signal conditioning, so that the ADC (analog to digital converter) can be set for the dynamic part of the signal. In most cases this should be set to AC. However, this unconditionally introduces a first-order, high-pass filter, which is usually specified as a cutoff frequency at, typically, 1 Hz. It is important to realize that a cutoff frequency as specified by manufacturers is a –3 dB limit, which is to say, the signal is attenuated by approximately 30% at this frequency! The amplification as a function of frequency follows:

$$\frac{U_{\text{out}}}{U_{\text{in}}} = \frac{f/f_c}{\sqrt{1+\left(f/f_c\right)^2}} \qquad (1)$$

where f_c is the cutoff frequency in Hz. From Eq. 1 it follows, that a 1% error requires the frequency to be above approximately seven times the cutoff frequency.

IEPE sensors are the most commonly used sensors today. Almost all data acquisition devices for noise and vibration signals are equipped with a built-in current supply for such sensors. A warning should be issued against unwarily using such built-in supplies! For various reasons, instrument manufacturers want to keep the IEPE currents as low as possible and it is not uncommon to use the minimum specified by the IEPE standard – 2 mA as the current feed. Also, the voltage used to produce the current feed is often kept around the minimum of 18 V. This does not to work well with many sensors on the market. Current should be at least 4 mA and the driving voltage at least 24 V to avoid serious problems with sensor behavior. Longer cables will require even more current and large output signal voltages could require higher driving voltage.

TEDS (transducer electronic data sheet) sensors and data acquisition equipment that supports such sensors, are becoming increasingly popular. TEDS sensors are standardized by IEEE 1451.4. This standard allows software to read out important data such as sensor type and make, sensitivity, etc. out of a memory chip in the sensor. Naturally this is very desirable and should be seriously considered when purchasing new equipment today, since one of the most time consuming parts of our measurements is the “book keeping.”

Sampling and A/D Conversion. The next step in the signal path is an adjustment of the signal level to the antialias filter and ADC (analog to digital converter), which is usually controlled in software by selecting a particular input voltage range. After this, the signal is filtered by a low-pass filter before being converted to a digital signal in the ADC. An analog antialias filter is absolutely necessary for most types of signals encountered in noise and vibration measurements. Still, it is surprising how often we find custom made equipment where the analog antialias filter has been omitted.

Today, ADCs in noise and vibration measurement systems are almost exclusively sigma-delta converters (sometimes called delta-sigma converters). Essentially, the sigma-delta converter consists of the three elements inside the dotted box of Figure 1. Sigma-delta converters are based on delta modulation, which is a technique where the analog value of the signal is represented by the average of the digital signal. The A/D converted signal is such that if the digital signal is represented with the numbers –1 and +1, then the *average* of the digital signal from time zero to any instant t equals the analog signal value at time t. The initial digital signal is only one bit in resolution, but is then filtered through a digital filter. Such a filter performs a weighted summation, which has two implications: (a) it averages a number of input values to produce an output value, and thus reduces the variance of the signal (i.e. increases the signal-to-noise ratio, SNR); and (b) it reduces the bandwidth of the signal. With decreasing frequency range, the effective number of bits increases. This is how manufacturers today can claim they are using 24-bit ADCs. They are using 1-bit ADCs, but are then filtering the signal down to a lower sampling frequency which increases the SNR to the same value as would be achieved by an ADC with 24 bits.

Sigma-delta ADCs provide almost ideal performance for spectral analysis. However, the design of the digital filter is important. For frequency analysis, this is never any problem. But for time or probability analysis, the digital decimation filter succeeding the A/D in Figure 1 has to be implemented as a linear-phase filter. This is not always the case in commercial systems today and should be checked by the user. As most software used today allows time data to be recorded (which is also our recommendation), it is tempting to think that the signal is validly sampled for any type of analysis, but this is not always the case! Perhaps it should be mentioned that it also matters if the digital filters are implemented with integer or floating-point precision. Integer arithmetic DSPs are faster and more inexpensive, but much care must be taken in the design, or the mentioned advantages above may be seriously deteriorated.

The use of 24-bit ADCs is desirable. Provided the analog electronics, etc. have matching low noise floors, the full dynamic range of the sensors can be retained with much less emphasis on adjusting proper input voltage range to the actual signal.

The importance of simultaneous sampling across a multichannel measurement system is probably well comprehended by most users. Perhaps a lesser known fact is the need for very accurate sampling intervals when performing frequency analysis. The discrete Fourier transform (DFT) is very sensitive to errors in the time interval between two samples. Time jitter in the sampling intervals causes a reduction in dynamic range in the spectrum, and roughly the standard deviation of the time interval error must be of the same order as the desired dynamic range.

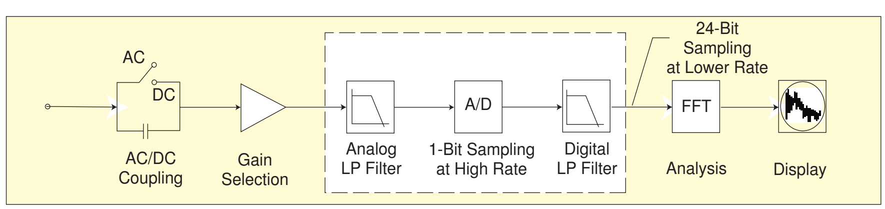

Figure 1. Schematic illustration of an FFT analyzer or a data acquisition system using a ΣΔ ADC for noise and vibration analysis.

This means that to obtain a 120-dB dynamic range at a 1-kHz sampling frequency, the standard deviation of the time jitter must be less than a nanosecond! If sample-and-hold circuits are used, this is usually not a problem, but in the case of home-built data acquisition systems, we have seen this often being neglected.

General Input Parameters. The concept of *dynamic range* is essential for frequency analysis, especially for frequency response measurements that often cover a very large range. Dynamic range is defined as the ratio of the largest to the smallest signal that can simultaneously be resolved by a measurement and is stated in dB. In most modern data acquisition hardware, the dynamic range is limited by spurious narrow-band signals (tones) that occur from clock frequencies, etc. in the hardware. In such cases, the dynamic range can be measured rather easily by shorting the input – for example connecting a 50 Ω BNC terminating plug to the input connector. After setting the input voltage range to 1 V, a spectrum is measured using a few ensemble averages to obtain a stable spectrum of the background 'noise' (which is not really noise but assumed to be periodic). The SNR in dB is then computed by reading the root mean square (rms) value of the highest peak in the spectrum, and using Eq. 2, where 0.707 is the rms of the full scale voltage. The spectrum will now contain only the spurious 'noise,' since the input is shorted, and the reading of the highest peak in the spectrum is the dynamic range. If the dynamic range is limited by random noise, it can alternatively be measured by calculating the rms value of the signal of the shorted channel (i.e. the background noise), and then calculating the SNR by:

$$\mathrm{SNR} = 20\log_{10}\left(\frac{0.707}{\mathrm{rms}}\right) \tag{2}$$

Channel-to-channel cross-talk is a measure of how much contamination from surrounding channels leaks into a measurement channel. If the cross-talk is lower (in dB) than the dynamic range of the instrument, it will be the factor that determines the useful dynamic range of the instrument. In fact, this is sometimes the case for existing instruments on the market. This factor is relatively easy to measure by connecting a sinewave generator with a signal close to the full-scale voltage of all channels (that is, all channels are set to the same range), and with a suitable frequency, to one channel. The remaining channel inputs are shorted, and spectra of all channels are measured for dynamic range as described above. The sine tone will normally stick up through the noise floor of at least some channel, and the cross-talk is the dB value of the ratio of the spectrum peak to the full-scale voltage.

Cross-channel match is an important measure for two-channel measurements such as frequency responses. It measures the amplitude and phase match of the entire signal chain on the two channels. To measure this, the same random noise signal is connected to both channels under investigation, and the frequency response between the two channels is calculated. The amplitude and phase curves obtained are the cross-channel match, usually measured in percentage and degrees, respectively.

Data Recording and Analysis. Digital noise and vibration measurement systems have evolved considerably since the early 1960s. Due to the performance limitations of ADCs and computers, data had to be recorded on analog tape recorders, then played back usually at much lower speed for the ADCs to cope with the signals. FFT analyzers in those days were unknown, so a spectrum would be computed by a discrete Fourier transform (DFT), at a speed we do not wish to think about today.

Today, hard disk storage is so inexpensive that, for most applications, time data can be stored at an economical cost. However in the 1970s, when the first FFT analyzers were developed, this was not the case. For a few decades, it was necessary to compute averaged spectra in the frequency domain and discard time data. This had some definite drawbacks, as the parameters used for the FFT, such as frequency range, time window and FFT record length, determine the meaningfulness of the results. In many cases it was desirable to be able to change the FFT settings in the analysis phase, but this was impossible at the time. We find that many of today's analysis systems have not really been adapted for this new situation. It is possible to record time data, but many systems are not optimally designed for this procedure. Many users still average spectra and discard their time data for no good reason.

We advocate always recording time data, unless there is a very good reason not to, such as in some production testing systems or for some volume applications where very many similar products are to be measured and analyzed with a predefined procedure. Other reasons can be that it is desirable to immediately review frequency-domain information, such as frequency response functions and coherences. Although it is desirable to record time data in some cases, many of the measurement systems on the market today do not support this simultaneously with computing those functions. Having access to the raw time data, however, makes it possible to analyze data with different frequency resolution and time windows, etc. and this can add very much in the analysis results.

When it comes to software for analysis of noise and vibration signals, there are some requirements that should be considered. Besides recording time data, it is desirable to be able to control the analysis procedure by a programming sequence. Naturally, the recording system should also allow for plentiful documentation of the measurement situation, the measurement object itself, sensors, etc. We also advocate that menu systems should be avoided when it comes to the analysis procedure, as the documentation of the procedure used is difficult. A sequence of commands for the analysis procedure directly produces traceable documentation of the analysis that is much more useful.

A common "buzz word" heard today is open software. One of the most important protocols in an open software environment is open file formats, which secure the ability to access data presently and in the future. Therefore, it is surprising to us that so many of the systems on the market today do not offer documentation of the file formats. Over the years we have found numerous companies paying excessive amounts of money to convert their files from an old format to a new format when replacing their measurement systems. This should really not be necessary when vendors talk about open software!

Measurement Guidelines. Much can be said about how to properly make good noise and vibration measurements. We conclude this article by suggesting a few guidelines for the engineer to follow.

Check your sensors more often than you think might be necessary. If someone dropped an accelerometer on a concrete floor, it may still work on the calibrator, but it may be way off calibration at some frequency within the operating range. If you do not already have an impact hammer and a calibration mass, you should get these. Making a quick and simple measurement on the mass can reveal if the sensor is operating correctly over the entire frequency range of interest.

Always document everything that happens during your measurement. Record date, time and place of the measurement. Record measurement conditions, hardware used by serial numbers, etc. Always record more data than you think is necessary. Putting it another way: always record as much data as you think you can justify. You never know if you may have to downsample data to a lower frequency range, which reduces the number of samples accordingly. If you record data for some particular analysis, perhaps someone later wants to reanalyze it for some other purpose? These guidelines are especially important for transient data. Recording the signal for a considerable time before and after the actual event makes it possible to analyze if the background signal changes during the event. This can cause analysis problems.

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