I Data Acquisition

Chapter I-2

Analog data is acquired in digital form for any or all of the following purposes:

Storage

Transmission

Processing

Display

Data may be stored in either raw or processed form; it may be retained for short, medium or long periods. It may be transmitted over long distances (for example, to or from outer space), or short distances (from a lab bench to a minicomputer alongside it). The data may be displayed on a digital panel meter, or as part of a cathode-ray-tube presentation.

Processing can run the gamut from simple comparison to complicated mathematical manipulations. It can be for such purposes as collecting information, converting data to a useful form, using the data for controlling a process, performing repeated calculations to dig out signals buried in noise, generating information for displays, simplifying the jobs of warehouse employees, controlling the color of paint, the thickness of a wrapper, the speed of a subway train.

But it all starts with getting the data in digitial form, as rapidly, as accurately, as completely, and as cheaply as necessary.

The basic instrumentality for accomplishing this is the analog/digital converter; it can be a simple shaft digitizer, a DPM with digital outputs, or a sophisticated high-resolution high-speed device. To accommodate the input voltage to the specified conversion relationship, some form of scaling and offsetting may be necessary, performed with an amplifier (/attenuator). To convert analog information from more than one source, either additional converters or a multiplexer may be necessary. To increase the speed with which information may be accurately converted, a sample-hold is desirable. To compress an extra-wide analog signal range, a logarithmic amplifier may be found useful.

The properties of the data-acquisition system depend on both the properties of the analog data itself, and on what is to be done with it.

In this chapter, we shall show some of the configurations that have proven useful and/or popular and discuss some of the considerations involved in the choice of configuration, components, and other elements of the system. Additional information can be found in the chapters on the individual devices.

THEN AND NOW

Two decades ago, A/D converters capable of 0.05% performance and 50,000 sample/second conversion rates, cost about $8000, consumed about 500 watts, and occupied perhaps one-third of a cubic meter. Today, Analog Devices' AD7570 requires less than 20 microseconds for a 10-bit 0.05% conversion, lists for $49 in 100's, and is packaged in a 28-pin DIP. And it is designed for easy interfacing with the modern microprocessor.

In the past 20 years, through several "generations" of equipment, data-acquisition hardware has changed radically, thanks primarily to the semiconductor revolution, and prices, have come down to the point where digital, rather than analog, "massaging" of information is a matter of routine, rather than exotic necessity.

What have not changed, however, are the fundamental system problems confronting every digital data-systems designer. Of course, it helps to have small, quiet, low-cost, cool, low-drain components. But (s)he is still up against the laws of Mother Nature, who often seems to prefer to keep her secrets safely obscured by noise, rfi, ground loops, power-line pickup, and transients coupled into signal lines from machinery. Separating the signals from these obscuring effects, then, becomes a matter for ingenuity and imagination, coupled with a great deal of experience; it is not merely a matter of purchasing fast, high-resolution A/D converters. (But having them available at realistic cost provides incentives for giving them useful jobs.)

ENVIRONMENT AND COMPLEXITY

Data-acquisition systems can be separated into at least two basic categories: those suited to favorable environments (electrically quiet laboratories), and those intended for hostile environments (factories, vehicles, military surroundings, and remote installations). The latter group includes industrial process control systems where, for instance, temperature information developed by thermocouples located on tanks, boilers, vats, pipelines, bearings, oil burners, etc., (that are often spread over miles of factory real estate) is fed into a central computer that provides real-time process control. Included are digital control of steel mills, automated production processes, numerically-controlled machine tools. Any or all of these applications may be characterized by the vulnerability of data signals to the phenomena mentioned above and the requirement for almost routine isolation and measurement of off-ground voltages. Also included are *electrically*-noisy environments, such as generating stations, where thermocouples measuring bearing temperatures of rotating machinery are exposed to volts of interference caused by megawatt changes in load; and aircraft control systems, radar stations, etc.

On the other hand, for laboratory-instrument applications, and such test systems as those gathering long-term drift information on arrays of zener diodes undergoing constant-temperature life tests in well-shielded ovens, or gas chromatographs, automatic weighing machines, mass spectrometers, and other sophisticated instruments, the system designer's problems are related more to the performing of sensitive measurements (usually under favorable conditions) than to the gross problems of protecting the integrity of analog data.

Systems existing in hostile enviroments may require devices capable of wide temperature-range operation, excellent shielding, considerable design effort aimed at eliminating common-mode errors and preserving resolution, conversion at early stages, redundant paths for critical measurements, and (perhaps) considerable processing of the digital data to ensure that it is reliable. Measurements in the laboratory, with narrower temperature ranges and less ambient electrical noise, may be easier to make and communicate, but higher accuracies (or resolutions) may require more-sensitive devices, and a still-considerable degree of effort to preserve appropriate signal/noise ratios.

KEY FACTORS

The choice of configuration and circuit building blocks in data acquisition depends on several critical considerations:

1. Resolution and accuracy
2. Number of analog channels to be monitored
3. Sampling rate per channel
4. Throughput rate
5. Signal-conditioning requirements
6. The cost function

Besides the choice of appropriate component performance levels, careful analysis of the above factors is required to obtain the lowest-cost circuit configuration. Typical configurations include:

1. Single-channel possibilities
   Direct conversion
   Preamplification and direct conversion
   Sample-hold and conversion
   Preamplification, sample-hold, and conversion
   Preamplification, signal-conditioning, and any of the above
2. Multi-channel possibilities
   Multiplexing the outputs of single-channel converters on a processor bus
   Multiplexing the outputs of sample-holds
   Multiplexing the inputs of sample-holds
   Multiplexing low-level data
   More than 1 tier of multiplexers

Some of the more-interesting signal-conditioning options include

1. Ratiometric conversion
2. Wide-dynamic-range options
   High-resolution conversion
   Range biasing
   Automatic gain switching
   Logarithmic compression
3. Noise-reduction options
   Filtering
   Integrating converters
   Digital processing

Finally, in evaluating tradeoffs, there are at least three types of "budgets" that should be considered: cost budget, system time budget, and error budget.

SINGLE-CHANNEL CONVERSION SUBSYSTEMS

Direct Conversion

Figure 1 represents the simplest digitizing system, a lone A/D converter, performing repetitive conversions at a free-running internally-determined rate. It has power inputs and an analog signal input. Its outputs are a digital code word, including "over-range" indication, (in parallel or byte-serial form), polarity information (if necessary), plus a "status" output, to indicate when the output digits are valid.

Perhaps the most well-known converter of this kind is the digital panel meter, which consists of a basic A/D converter and a numerical display. For many applications, the sole purpose of digitizing is to obtain the numerical display, i.e., to use the DPM as a meter, rather than as a system component.

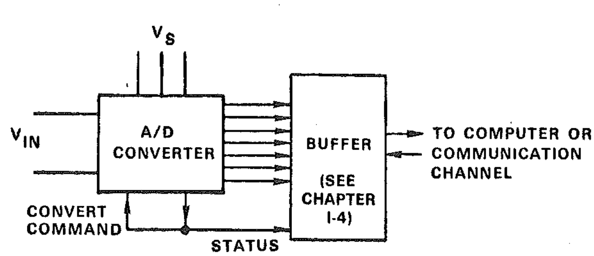


Figure 1. Simplest data-acquisition system configuration.

The DPM, however, is not necessarily the best way to digitize a single channel. Its two major shortcomings are: it is slow, and its BCD digitial coding must be changed to binary if its output is to be processed by binary equipment. When free-running, its output is strobed into an available interface register when the data is valid, rather than by an interface command.

Converters designed for system applications (including many DPM's, such as the AD2008) can usually receive external commands to convert or hold. For dc and low-frequency signals, the converter is usually a dual-slope type, which has the advantage that it is inherently a low-pass filter, capable of averaging out high-frequency noise, and nulling frequencies harmonically-related to its integrating period. (For this reason, the integrating period is usually made equal to the period of the line frequency, since the major portion of system interference usually occurs at that frequency and its harmonics.)

If the converter is responding to individually-important samples of the input, the maximum rate of change of the average input for full resolution, and the conversion rate, have the following relationship (binary conversion):

$$\left.\frac{dV}{dt}\right|_{max} = 2^{-n} V_{FS} / T_{CONVERT}$$

For example, if $V_{FS} = 10V$, $n = 11$ (1/2048 resolution) and $T_{convert} = 0.1s$, the maximum rate of change of input is about 1/20 V/s. At faster rates of change, 1 LSB changes cannot be resolved within the sampling period.

If, on the other hand, individual samples are not important, but large numbers of samples are to be dealt with, essentially delineating a stationary process, the only requirement is that the signal be sampled at least twice per cycle of the highest frequency of interest. For this purpose, in the example given above, the maximum signal frequency that can be handled is 5Hz.

So far, the context has been that of the dual-slope integrating A/D converter, which spends about 1/3 of its sampling period performing an integration, and the remainder of the time counting out the average-value-over-the-integrating-period as a digital number, and resetting to initial conditions for the next sample. It should be noted that the dual-slope type will always read out the average value, which results in a valid sample of the input waveform over the integrating "window."

Though it is slow, an integrating A/D converter, such as the quad-slope AD7550, is quite useful for measurements of temperature, battery discharge, and other slowly-varying voltages, especially in the presence of noise.

However, by far the most popular type of converter for system work is the successive-approximations device (described amply in Part II), since it is capable of high resolution (e.g., 16 bits), high speed (e.g., 1μs for 10-bit conversion), and quite reasonable cost. In the above example, if $T_{convert}$, using a successive-approximation converter, is 10μs, the maximum allowable dV/dt for maintenance of bit-at-a-time resolution becomes 500V/s, an improvement, but far from sensational.

The successive-approximation converter has the weakness that at higher rates of change, it generates substantial linearity errors, because it cannot tolerate change during the weighting process. The converted value lies somewhere between its values at the beginning and the end of conversion; but the time uncertainty approaches the magnitude of the conversion interval. Figure 2 illustrates this point. Finally, even if the signal is slow enough, noise rates-of-change (perhaps introduced by the signal itself) that are excessively large will cause erroneous readings that cannot be averaged, by either analog or digital means. An external sample-hold can greatly improve matters, as will be shown.

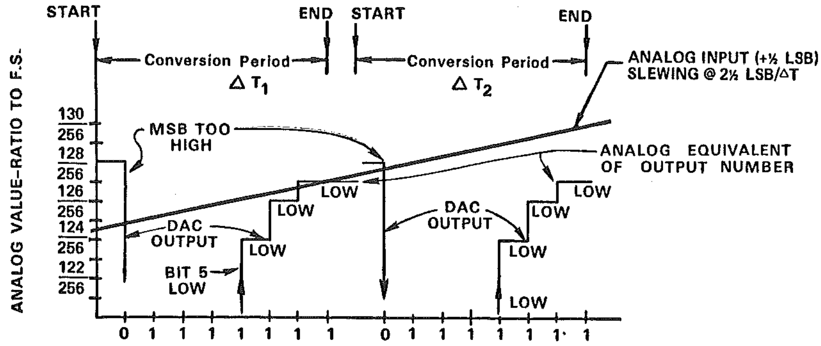


Figure 2. Successive-approximations converter error with fast-changing input (2½LSB per conversion interval). Note that output reading of 127/256 FS is the same for both intervals, though one starts at 124/256 and the other ends at 129/256. Error increases with ramp speed.

Direct conversion, especially near the signal source, is most useful if the data must be transmitted through a noisy environment. This can be seen, even in the case of an 8-bit converter (1/256 resolution) and a high-level 10V signal, if one considers that bits will be lost if the peak-to-peak noise level induced in the analog signal is greater than 40mV (approximately 10V/256); the standard TTL digital noise immunity, on the other hand, is 1.2V (or 2.0 – 0.8V), a gain in signal-to-noise of better than 10:1.

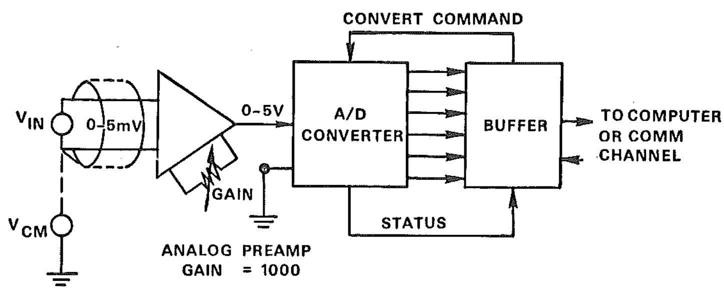


Figure 3. A/D Converter with preamplifier.

Preamplification and Direct Conversion (Figure 3)

Converters designed for OEM system applications are, in most cases* "single-ended" in reference to signal ground, and have normalized analog input ranges of the order of 5 or 10 volts, single-ended or bipolar. It makes sense to scale signal inputs up or down to the standard converter input level, to make fullest-possible use of the converter's available resolution.

If the signals are of reasonable magnitude (already preamplified or outputs of an analog computer), and already exist within the system, the scaling may be simply accomplished with operational amplifiers in a single-ended or differential configuration. If the signals are from outside the system, or are quite small, or have an appreciable common-mode component, a differential instrumentation amplifier may be profitable used, with characteristics depending on the gain required, the signal level, the needed CMR, bandwidth, impedance levels, and cost tradeoffs.

Finally, if the input signals must be galvanically isolated from the system, a light- or transformer-coupled isolation amplifier must be used to break all conductive signal paths. This kind of isolation is essential in some medical-instrument applications. It is also useful where large common-mode spikes are encountered, for industrial applications requiring *intrinsic safety,* and for all applications in which the signal source is at a high off-ground potential.

*Extremely-high-resolution converters, such as the ADC-16Q, have a high-CMR differential front end. DVM's, and some DPM's, also tend to have differential inputs, in consonance with their low full-scale-input levels (e.g., 0.1999V).

Sample-Hold and Conversion (Figure 4)

A successive-approximations converter can be made to operate at considerably-greater accuracies at high speeds, overcoming the weaknesses mentioned above, by introducing a sample-hold at its input. Between conversions, the sample-hold acquires the input signal, and, just before conversion takes place, it is placed in *hold,* where it remains throughout the conversion. It can be seen that, if the S/H responds instantaneously and accurately, the converter can convert changes (from the preceding sample) of any magnitude accurately, at speeds up to the conversion rate. In practical sample-holds, however, there will be acquisition time, tracking delay, and aperture time; typical values of these quantities are 2μs to 0.01%, a fraction of a μs, and 25ns, with 2-3ns uncertainty. If the aperture time and tracking delay compensate one another, or are unimportant as long as they are consistent, the principal source of time error is the aperture *uncertainty.* The relation between aperture uncertainty and maximum rate of change for maintaining resolution in an n-bit system is

$$dV/dt\Big|_{max} = 2^{-n} V_{FS}/t_{apu}$$

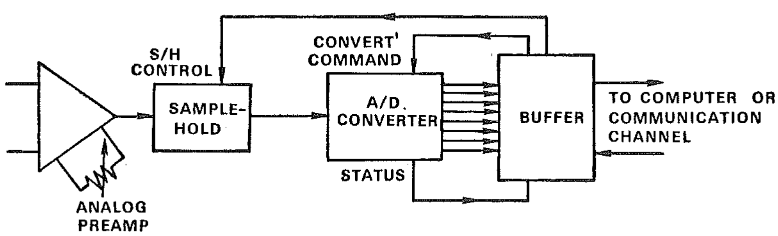
for the example given, $(f_{sample} = 100kHz,\ V_{FS} = 10V,\ t_{apu} = 3ns)$

$$dV/dt\Big|_{max} = 5mV/3ns = 1.67V/\mu s$$

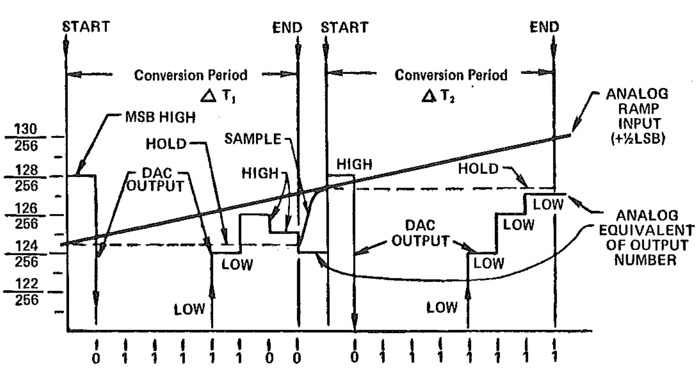
This number is also limited by the *slew rate* specification of the sample-hold.

Figure 4b shows, in contrast with Figure 2, that the successive-approximations converter, with a constant input applied by the S/H, will deliver an accurate digital representation of the beginning input at the end of each conversion interval. Any errors that are functions of time will be due to errors of the sample-hold, including the acquisition errors above, plus droop during the conversion interval, and any linearity, offset, and transient errors. Noise present on the signal, though sampled and converted, may be susceptible to digital averaging by the processor.

Since sample-holds usually operate at unity gain, with errors referred to full-scale (which should be the same as the converter's full scale range), scaling or preamplification should occur before the signal is applied to the sample-hold.



4a. Sample-Hold in single-channel data acquisition.



4b. Successive-approximations converter example of Figure 2, preceded by sample-hold. Digital output is within ½LSB of analog input at start of conversion, 124/256, and 127/256.

Sample-hold devices can be used with other types of converters, to establish precise timing of the signals being sampled, independently of the time required by a given device to complete a conversion. Their utility is especially evident if the conversion time is variable, as in counter types.

Signal Conditioning (Figure 5)

This is a blanket term that includes a wide variety of possibilities. Scaling of input gains to match the input signal to the converter's full-scale range is a simple, obvious example. One might also include dc offset to bias odd ranges, such as 2.5 to 7 volts, to levels more compatible with standard converters. Preamplification, as discussed earlier, is a typical example. Linearizing of data from thermocouples and bridges can be performed by analog techniques, using either piece wise-linear approximations (generated by biased diodes) or smooth series-approximations using low-cost I.C. multipliers; it could be done digitally, after conversion with a microprocessor or by using a ROM to store the inverse function.

Analog differentiation could be used to measure the rate at which the input varies; integration could be used to obtain total dosage from a rate of flow. Either could be used to produce a 90° phase shift; an op amp could be used to provide an arbitrary phase shift. Sums and differences can be used to reduce the number of data inputs (analog data *reduction*).

Analog multipliers could be used to compute power by squaring voltage or current signals, or by multiplying them together. Analog dividers of various types could be used to compute ratios or the logarithms of ratios, or square roots. Devices that compute $Y(Z/X)^m$ can take ratios over wide dynamic ranges, and perform ideal-gas computations.

Comparators can be used to make decisions based on analog levels (e.g., to convert only when an input exceeds a threshold or is within a "window.") Op amps and diodes may be used to perform simple "ideal diode" functions.

And – what seems almost like getting "something for nothing" – logarithmic modules can be used for range compression to permit the conversion of signals having resolutions of 10^6 with 12-bit converters. (This will be discussed later in the chapter.)

Active filters are essential elements to minimize the effects of noise, carriers, and unwanted high-frequency components of the input signal. Their growth of use, and the increase of interest in their design are reflected in the large number of magazine articles and the preponderance of filter discussions in such publications as the IEEE *Transactions on Circuit Theory*, plus a growing number of books on both analog and digital filtering.

One could go on and on, but the basic point should have been made: That in system design, all data-processing need not be digital (with all due respect to the great potential inherent in the use of microprocessors). Analog circuits can perform processing or data reduction effectively, reliably, and economically, and should be considered as alternative ways of reducing numbers of transmission channels, software complexity, noise, and – more often than not – cost.

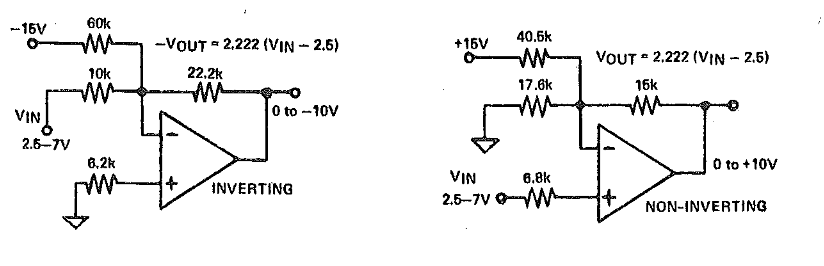


Figure 5a. Op amps used for offset and scaling.

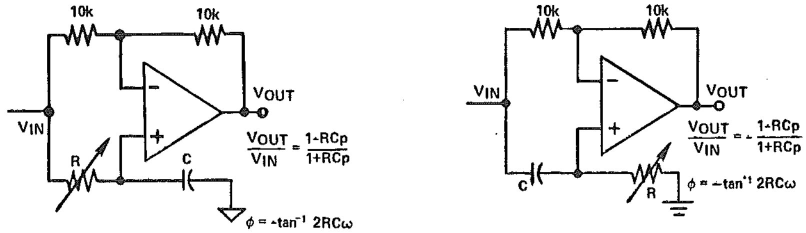


Figure 5b. Op amp to generate arbitrary phase shift, Gain = 1, all-pass.

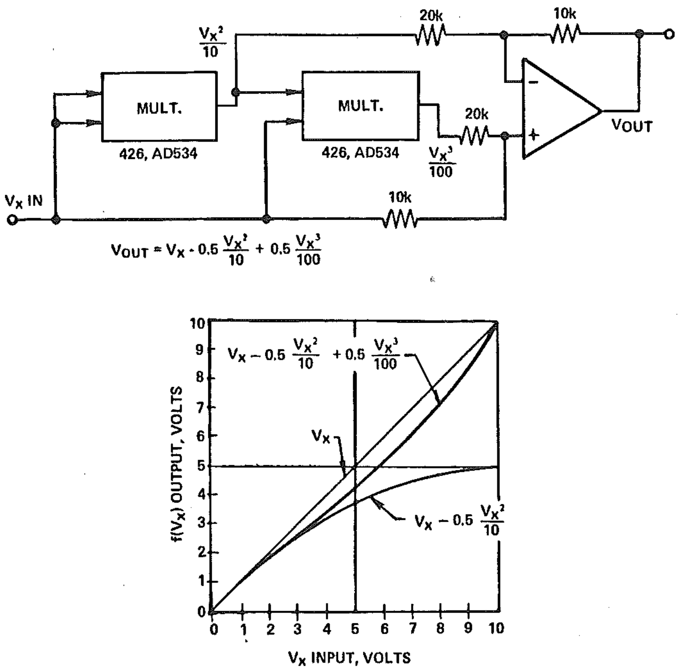


Figure 5c. Using multipliers for nonlinear functional realtionships (such as linearization).

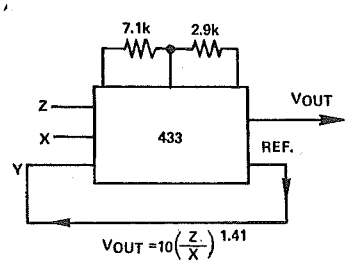


Figure 5d. Ratio of two voltages raised to arbitrary non-integral power.

MULTI-CHANNEL CONVERSION

In multi-channel conversion systems, elements of the acquisition chain may be shared by two or more input sources. This sharing may occur in a number of ways, depending on the desired properties of the multiplexed system. Large systems may combine several different kinds of multiplexing, as well as cascaded tiers of the same kind.

Multiplexing the Outputs of Single-Channel Converters

Although the conventional way to digitize data from many analog channels is to introduce the time-sharing process, whereby the input of a single A/D converter is multiplexed in sequence among the various analog sources, an alternative parallel conversion approach is becoming increasingly practicable. Cost of A/D converters has dropped radically during the past decade, and it is now possible to assemble a multi-channel conversion system, with the seeming extravagance of one converter for every analog source, just as economically as the conventional analog multiplexed system. (Figure 6.)

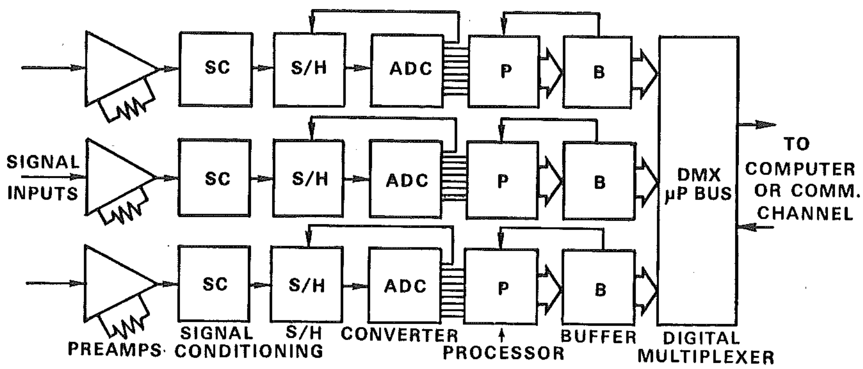


Figure 6a. Basic multi-channel conversion scheme, using digital multiplexing before transmission.

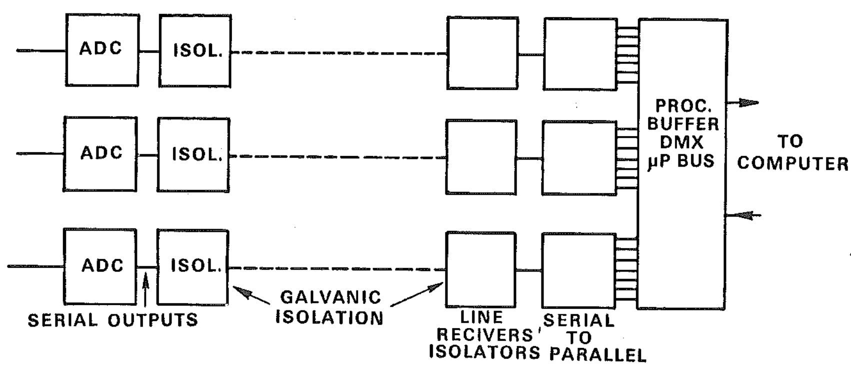


Figure 6b. Multi-channel conversion using remote A/D converters.

There are a number of important advantages to this parallel conversion approach, which, by the way, is virtually standard practice for resolver/synchro conversions beyond about the 10-bit accuracy level. First of all, quite obviously, slower converters may be used to obtain a given digital throughput rate; alternatively, the converter-per-channel may run at top speed, providing a much greater flow of data into the digital interface. For a constant data rate, however, with more channels (and fewer conversions per channel), the reduced conversion speed, plus the fact that each converter is looking at continuously-changing data, rather than jumping from one level to another may allow the sample-holds to be eliminated, at a cost saving. Fewer conversions also mean that a slower converter might be used, generally resulting in even further cost saving, especially since some channels may not require a great deal of resolution.

The parallel-conversion approach provides a further advantage when applied to industrial data-acquisition systems, where many strain gages, thermocouples, thermistors, etc., are strung out over a large geographical area. In essence, by digitizing the analog signals right at their source and transmitting serial digital data, rather than the original analog signals, back to the data center, a considerably immunity to line-frequency (50-60-400Hz) pickup and ground-loop interference is achieved. Among other factors, the digital signals can be transformer or optically coupled, for example, to gain complete electrical (hence ground loop) isolation. Also low-impedance-digital drive and receiving circuits drastically prune vulnerability to noise.

Not least, among the subtle benefits of digitzing sensor signals at their source, is the ability to perform logical operations on the digitized data before it is fed into the computer. In this way, for example, main-frame involvement with data is streamlined and redundancies are minimized. More-specifically, logic circuits can arrange to access data from slowly-varying thermocouple sensors less frequently, while reading-in data from critical sources at enhanced speed. In fact, the versatility of a digital subsystem may be exploited to make its own decision as to when a particular data channel should be fed into the computer: if certain signal sources remain constant or within a narrow range for long periods, then change rapidly later in the process, it is possible to ignore these data until the changes occur. (A local microcomputer can store the stationary values and make the decisions).

In summary, a great deal of flexibility and versatility is gained by transferring the interface process from analog multiplexing to digital multiplexing. Logic decision circuits can exercise judgement on when and what data to feed the computer, and, in general, can give the overall interface a much larger measure of autonomy than is possible with an entirely-analog conversion system. (The computer cannot make decisions about the data submitted by an analog multiplexing system until it has received the data upon which to base its judgements...this means that the data have been interfaced before the computer can decide that that particular piece of information is redundant. And there is no guarantee that it will be redundant on the next pass.)

Finally, it should be noted that if the data is being transmitted from a lunar vehicle to Earth, the channel is quite crowded, and the sort of *redundancy-reduction data compression* described above is absolutely essential to make sure that the items of data that get through are those having the highest priorities, by virtue of containing intelligence rather than redundant information.*

For each channel of the digitally-multiplexed system, there could be the array described earlier: preamplifier, signal-conditioning, sample/hold, converter. It is also possible that for one or more of the channels, there are a number of sequentially-multiplexed sub-channels, especially if they are carrying similar information.

Multiplexing the Outputs of Sample-Holds

Working back from the interface (with a minimum number of shared elements) towards the more-conventional situation in which the number of shared elements is maximized, we consider the case of a shared A/D converter, with a multiplexer at its input, switching among the outputs of a number of sample-holds (Figure 7). This configuration is found where sample holds are updated rapidly, even simultaneously, then read out in some sequence. It is generally a high-speed system, in which all items of data delineating the state of the system must do so for the same given instant. Multiplexing may be done

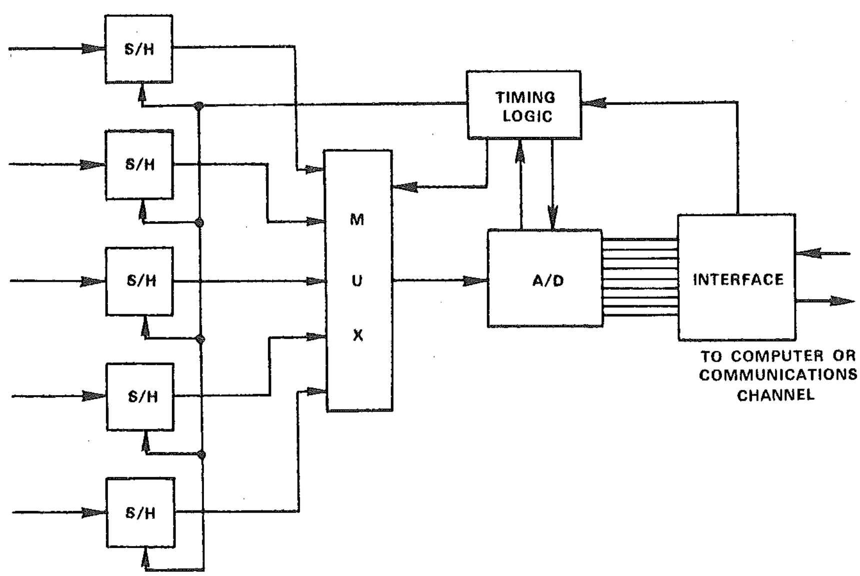


Figure 7. Multiplexed simultaneous sample system.

*"New approaches to Data-Acquisition System Design," by T. O. Anderson, *Analog Dialogue,* Vol. 5, No. 1, January, 1971.

sequentially, or by random addressing. For this kind of operation, the former is more typical. The sample-holds must have sufficient freedom from droop to avoid accumulating excessive error while awaiting readout, which period may be considerably longer than in the case of the converter-per-channel. Increased throughput rate could be obtained by using additional converters, with fewer multiplex switch points and faster update rate.

Applications that might require this approach include wind-tunnel measurements, seismographic experimentation, or in testing complex radar or fire-control systems. Often, the event is a one-shot phenomenon, and the information is required at a critical point during the one-shot event. . .such as, for example, when a supersonic air blast hits the scale model.

Multiplexing the Inputs of Sample-Holds

The next step towards increased sharing is to share the sample-hold as well as the A/D converter. Figure 8 shows the typical system embodying this idea. For most-efficient use of time, the multiplexer is seeking the next channel to be converted, while the sample-hold, in *hold,* is having its output converted. When conversion is complete, the *status* line from the converter causes the S/H to return to *sample* and acquire the next channel. Then, after the acquisition time is completed, either immediately, or upon command, the sample-hold is switched to *hold,* a conversion begins, and the multiplex switch moves on.

This system is slower overall than the previous example, and, since the channels may tend to be diverse rather than identical, the multiplexer could equally well be switching sequentially or in a random-access mode. For some systems, a manual mode, for checkout, may also be desired. In the random-access mode, it is quite possible that some channels (those with more "intelligence", i.e., change/time), will be accessed more frequently).

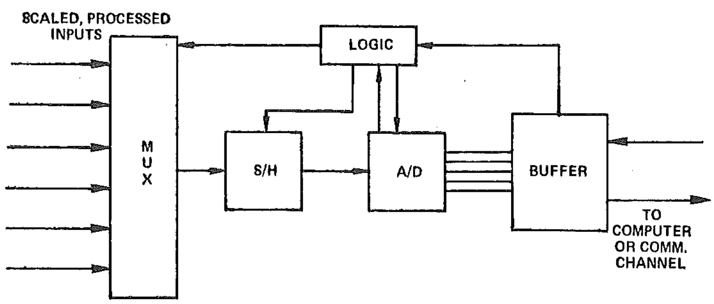


Figure 8. Multi-channel analog multiplexed interface.

Multiplexing Low-Level Data (Figure 9)

The idea here is that, in addition to sharing of the converter and the sample-hold, expensive instrumentation-amplifier capacity must also be conserved. The decreasing cost of instrumentation amplifiers (in fact, of amplifiers of all types), plus the disadvantages of low low speed and the engineering effort involved in ensuring the successful transmission and multiplexing of low-level data, are likely to result in decreasing use of this system approach. Some of the considerations are discussed in the chapter on Multiplexing.*

Low-level multiplexing is often involved with the use of programmable-gain amplifiers, or the even-more-sophisticated automatic range-switching preamps, which combine the use of converters having modest resolution with range-switching controlled from the interface to obtain additional significant bits. (For example, a 12-bit converter, and 32 steps of adjustable gain, can provide 17-bit resolution, assuming – big *if* – that the resolution is actually present in the signal and that the system is capable of handling it without degradation.)

All the difficulties that are inherent in single-channel low-level circuitry are compounded by the addition of low-level multiplexing of *n* such channels. Not only is guarding necessary, but individual channels should be individually guarded. And, following the principle

*Analog-Digital Conversion Handbook, Analog Devices, Inc., 1976, $3.95.

that the guard should be present right up to the input of the preamplifier, the guard too must be switched (or else driven from the common-mode level as measured separately at the amplifier*). Not only must the problem of pickup be considered, but the new dimension of crosstalk is added. And not only signal-to-signal crosstalk (not a great problem with small differential signals), but also common-mode-to-signal. Not only must input capacitance be balanced, but it must be balanced in the context of a multiplex switch having at least two circuits.

The saving grace (though perhaps not for some) is that systems of this sort are slow, and a large capacitor across each pair of input leads can reduce capacitive unbalance and high-frequency noise, without slowing-down the system excessively.

The usual low-level problems of thermal unbalance at connectors, lead junctions, and switches (and don't forget kovar-to-copper at IC's), must also be dealt with. All such unavoidable thermocouples should have thermal symmetry.

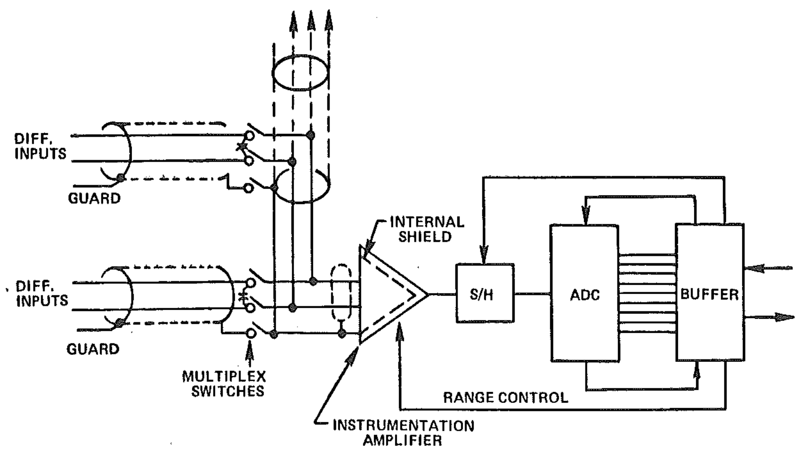


Figure 9. Low-level multiplexing.

More than One Tier of Multiplexers

If there are 64 channels to be multiplexed, the problems of stray capacitance (including capacitive unbalance) are worsened by the parasitic capacitance of the *off* channels on the conducting channel: if there are n channels, the capacitance will be $(n - 1)\ C_{DGO}$ plus the usual stray wiring capacitance. It is practical to reduce this capacitance by using two tiers of multiplexer. In the above example, with 8 channels per switch, the capacitance is reduced to $14C_{DGO}$ from $63C_{DGO}$.

SIGNAL CONDITIONING

Discussed here are a few topics that keep coming up in connection with data-acquisition systems.

Ratiometric Conversion

Some A/D converters have a *ratiometric*, or "external reference" connection, allowing the output digital number to represent the ratio of the input to an arbitrary (within specified limits) reference input. In effect, the device becomes an analog divider with digital readout.

Devices of this sort are useful in making precision measurements that ignore variation of a device reference. For example, Figure 10 shows how a potentiometer ratio can be measured, independently of variations of the applied voltage, by applying the same voltage to the reference input of the converter.

*The extra switch points can be eliminated by passively summing the voltages at the amplifier inputs, via high-impedance buffers, with two equal resistors, $CMV = \frac{1}{2}(e_1 + e_2)$, and buffering with a simple follower.

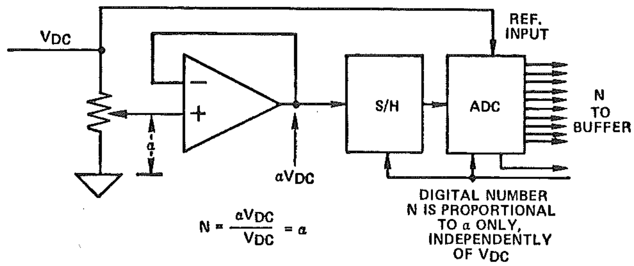


Figure 10. Measuring a potentiometer ratio, independently of the applied voltage, a ratiometric measurement. If the sensor is a bridge, the amplifier could be an instrumentation type instead of a follower.

In a multiplexed system, where measurements may be taken from a number of similar devices, such as strain-gage bridges, the common bridge supply may be used as the converter's reference to eliminate normal-mode gain error caused by supply-voltage variation.

Wide Dynamic Ranges

The need for wide-dynamic-range signal conditioning in a single channel may occur in two basic ways: Either it is necessary to resolve a voltage anywhere in the range to a high degree of accuracy, relative to full scale, (for example in the measurement of position in a follow-up system); or it is sufficient to measure a quantity having a wide range of variation to modest accuracy, relative to actual value (for example, to within 1%, over a 10,000:1 range).

For signals in the first category, a high resolution-and-linearity converter (such as the ADC-16Q) is the simplest answer. Another possibility is the use of a moderate-resolution converter (e.g., 12-bit resolution), preceded by an amplifier with switched gain (Figure 11) controlled from the digital interface. In one form of operation, a trial conversion is performed at the lowest gain; if the MSB is 0, the gain is doubled and another conversion is performed; if the MSB is still 0, the gain is doubled again, etc., until either the MSB is turned on or the top end of the range is reached. Each doubling represents an additional bit of resolution. This scheme may be programmed with the RTI1200 interface (Chapter I-4), for up to 15 bits of resolution (but not accuracy).

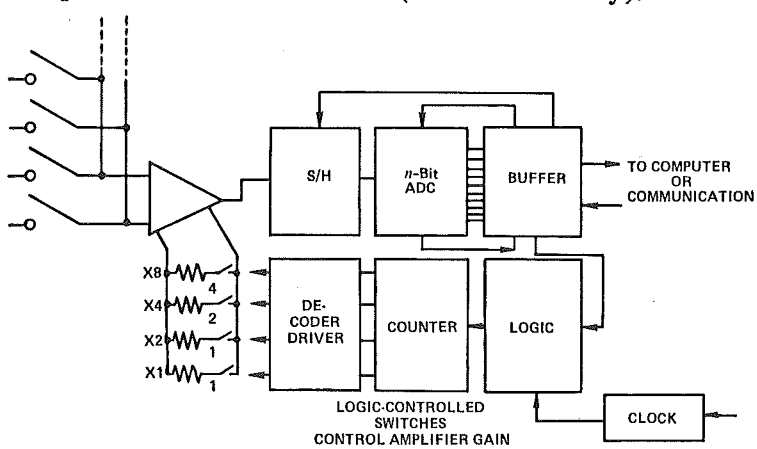


Figure 11. Switched-gain amplifier.

Yet another possibility, when seeking accurate measurements of small variations about a fixed value of voltage, is to sum a voltage equal to the nominal fixed value (and opposite in polarity) and measure the differences. If the voltage is applied (Figure 12) via a high-resolution DAC, the interface can keep track, digitally, of both the initial value and the difference voltage, using an ADC of quite modest performance. (The tradeoff here is the cost of a high-resolution DAC, such as the DAC-1138, plus logic and a modest 8-, 10- or 12-bit ADC, vs. a 16-bit ADC.)

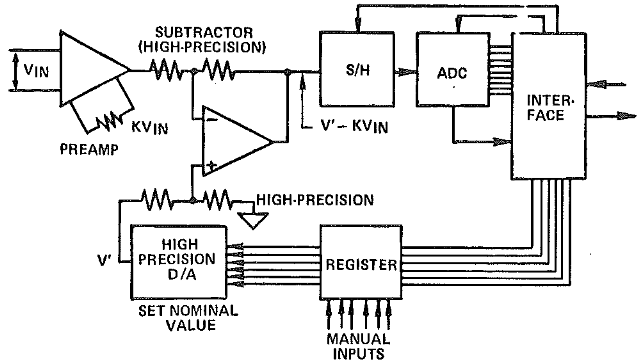
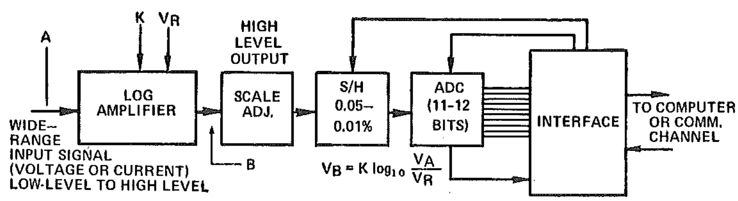


Figure 12. Use of high-resolution DAC to measure small deviations about a precisely-determined value.

In any of these schemes, it is essential to keep in mind that every element in the front end, wiring, preamplifiers, components, references, must be compatible with the resolution and accuracy sought. This also includes the noise level.

For signals in the second category, the switched-gain amplifier is a satisfactory, but slow-and-expensive approach. An intriguing alternative is to use a logarithmic amplifier for for data compression (Figure 13).

The error of a logarithmic amplifier, after calibration, is a *log conformity* error (nonlinearity on a semi-log plot) that is specified in terms of a maximum value at the output, or a maximum ratio to actual input over a specified range. For example, 1% log conformity error means that the error at the output, for 2V/decade* scaling, is 8.6mV, corresponding to an input uncertainty of ±1%. Typical input *voltage* range (i.e., for the Analog Devices 755) is 1mV to 10V. The corresponding output-voltage range is ±4V (i.e., ±2 decades at 2 volts per decade, with respect to a 0.1V reference level.) Since an error of 1% referred to the 1mV minimum input signal is $1/10^6$ of full-scale input, and since the corresponding output error of 8.6mV is $0.0086/8 = 1.075 \times 10^{-3}$ of the output swing, the dynamic range of the signal has been compressed by a factor of 1000, as a result of the logarithmic transformation. This means that a 12-bit converter (with suitable scaling) can be used to digitize the log amplifier output, with a quite-comfortable error margin.



A: INPUT, e.g. 10V to 1mV ±1% (10μV Resolution at Low End)
B: OUTPUT, ±4V ±8.6mV (K = 2, V_R = 0.1V) (Resolution About 0.1%)
If K Adjusted to Match ADC, "Scale Adj." Not Necessary

Figure 13. Low amplifier used for range compression in data-acquisition system.

Though it might appear that the representation of data having an inherent 20 bits of resolution ($10^6 \cong 2^{20}$) by a signal having 12-bit resolution is getting "something for nothing," in violation of *some* Natural Law, the scheme really works. There are, however, some points to consider:

1. Compression is achieved by exponentially distorting the relative value of the Least Significant Bit. Thus, for a 10,000-to-1 signal range, represented by ±4V output, an LSB (of 12 bits, offset binary, suitably scaled) is worth 23mV at 10V input (i.e., $10\ [1 - \log_{10}^{-1} 8/8192]$) and 2.3μV for 1mV input. Therefore, while the approach is quite useful for compressing data requiring essentially constant *fractional* error (e.g., 1%) anywhere in a

*A *decade* is a 10:1 range of input voltage or current.

wide range, it is not at all suited to applications requiring high resolution (e.g., 0.01% FS) *at any point* in the range.

2. Since the digital number is a logarithmic representation of the analog input signal, it must be dealt with as such in the digital process. If the number is to be used in computation, it should be antilogged, using a ROM and/or processor computing capacity, unless of course the computation is facilitated by the availability of a logarithmic relationship. If the data is simply to be stored or transmitted, and eventually returned to analog form unchanged, it does not require any further digital transformation, just an analog antilog operation following the output D/A conversion (unless logarithmic analog data is acceptable).

3. Since a logarithmic function is inherently unipolar (the logarithm is real only for positive values of the argument – positive signals require a 755N, negative signals a 755P), it is far from ideal for signals that are inherently zero-centered. While it may be useful to bias some types of input signals into a single polarity, functions that demand symmetrical treatment may be badly distorted by the wide variation, in both resolution and speed, between zero and full-scale input. Such functions would profit by a type of compression that is symmetrical about zero. An example of an easily-obtained form is a $\sinh^{-1}$ function (Figure 14), which involves two complementary antilog transconductors (752P and 752N) in the feedback path of an op amp. The resulting function is logarithmic for larger values of input, but it passes through zero, essentially linearly (but slowly).

Noise Reduction

Like diseases, noise is never eliminated, just prevented, cured, or endured, depending on its seriousness and the costs/difficulty of treating it.

Noise in data-acquisition systems takes three basic forms, *transmitted noise,* inherent in the original signal, *inherent noise,* generated within the devices used in data acquisition (preamps, converters, etc.) and *induced noise,* "picked up" from the outside world, power supplies, logic, or other analog channels, by magnetic, electrostatic, or galvanic coupling.

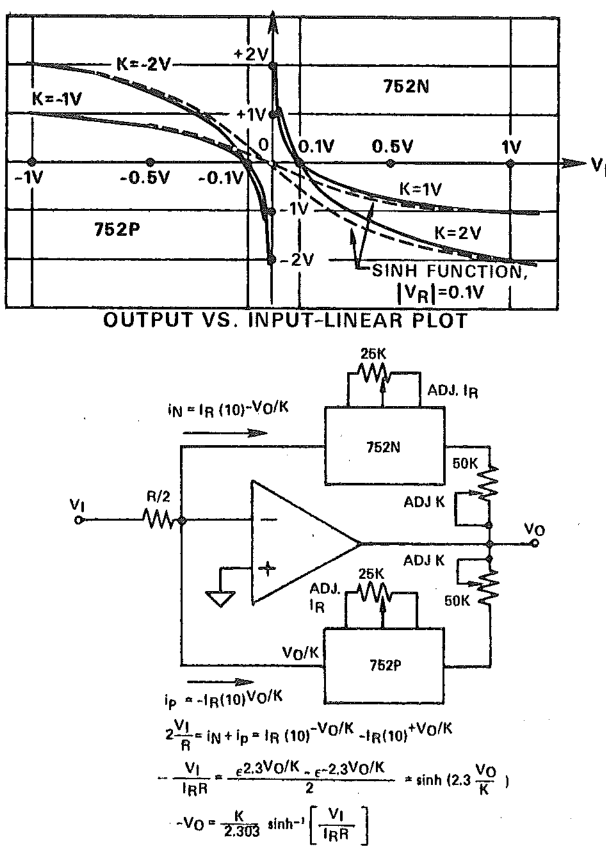
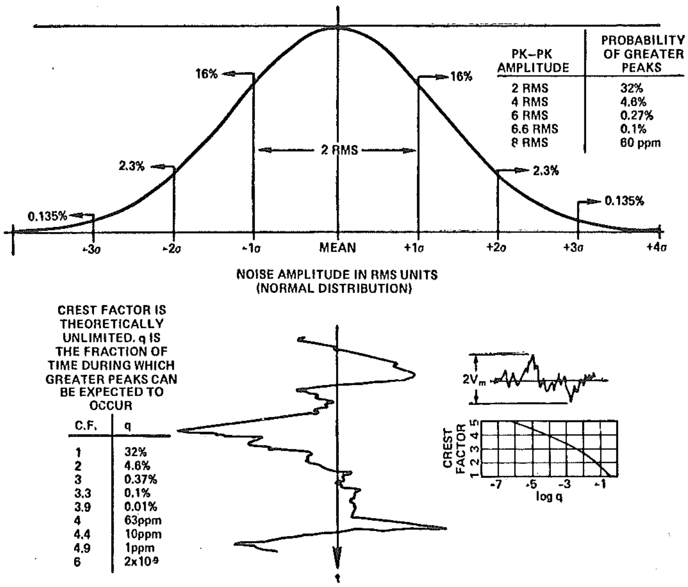


Figure 14. Bipolar signal compression using complementary logarithmic transconductors to synthesize $\sinh^{-1}$ function.

Noise is either *random* or *coherent* (i.e., correlated to some noise-inducing phenomenon within or outside of the system). Random noise is usually generated within components, such as resistors, semiconductor junctions, or transformer cores, while coherent noise is either locally-generated by processes, such as modulation/demodulation (e.g., chopper-stabilization), or coupled-in. Coherent noise often takes the form of "spikes," although it may be of any shape, including – collectively from many sources – pseudorandom.

Noise is characterized in terms of either *root-mean-square (rms)* or *peak-to-peak* measurements, within a stated bandwidth.* Random noise from a given source, within a given bandwidth, will give consistent *rms* measurements. For a typical gaussian amplitude distribution, and a sufficient number of measurements, one may expect a consistent relationship between the probabilities of obtaining peaks of given size in relation to the rms, as shown in the Tables in Figure 15.



PK–PK AMPLITUDE	PROBABILITY OF GREATER PEAKS
2 RMS	32%
4 RMS	4.6%
6 RMS	0.27%
6.6 RMS	0.1%
8 RMS	60 ppm

C.F.	q
1	32%
2	4.6%
3	0.37%
3.3	0.1%
3.9	0.01%
4	63ppm
4.4	10ppm
4.9	1ppm
6	2×10^{-9}

Figure 15. RMS vs. peak-to-peak amplitudes for gaussian noise.

RMS values of noise from uncorrelated sources (e.g., from different devices, or from different portions of the frequency spectrum of the same device) add as the square-root of the sum-of-the-squares. However, if noise is dominated by picked-up spikes, root-sum-of-squares is of small comfort.

As we have indicated at the beginning of the chapter, there are two basic forms of system-design problem: those involving essentially ordinary signal levels in unfavorable environments, and those involving extremely high-resolution measurements in favorable environments.

For unfavorable environments, where the major source of noise is *induced noise,* the designer must rely on early preamplification and conversion, isolation, shielding and guarding, signal compression and filtering, and – where possible – an information rate (via fast sampling or parallel paths) that has enough redundancy to allow the digital processor to retrieve data via correlation and summation.

In favorable environments, where the measurement process and the processing hardware introduce the major portion of the uncertainty, the emphasis must be placed on measurement techniques, filtering, choice of data-acquisition hardware for best resolution, and – again – the use of high-speed digital processing for signal retrieval, including drift compensation and scale-factor adjustment.

Where noise is likely to have large spikes as a major component, the integrating-type converter (dual-slope) usually provides additional filtering. For random noise, if there are sufficient samples taken of a given signal channel, the statistical properties of the noise are imparted to the digital output, which may be filtered by digital techniques.

*For a useful discussion of the properties of noise, see "Noise and Operational Amplifier Circuits," in *Analog Dialogue,* Vol. 3, No. 1.