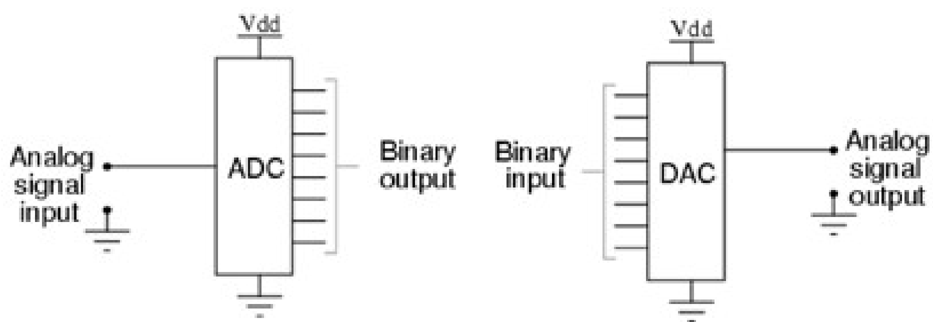
ANALOG TO DIGITAL CONVERSION

The natural state of audio and video signals is analog. When digital technology was not yet around, they are recorded or played back in analog devices like vinyl discs and cassette tapes. The storage capacity of these devices is limited and doing multiple runs of re-recording and editing produced poor signal quality. Developments in digital technology like the CD, DVD, Blu-ray, flash devices and other memory devices addressed these problems. For these devices to be used, the analog signals are first converted to digital signals using analog to digital conversion (ADC)**.** For the recorded audio and video signals to be heard and viewed again, the reverse process of digital to analog conversion (DAC) is used. ADC and DAC are also used in interfacing digital circuits to analog systems. Typical applications are control and monitoring of temperature, water level, pressure and other real-world data.

An ADC inputs an analog signal such as voltage or current and outputs a digital signal in the form of a binary number. A DAC, on the other hand, inputs the binary number and outputs the corresponding analog voltage or current signal.



Sampling rate

The analog signal is continuous in time and it is necessary to convert this to a flow of digital values. It is therefore required to define the rate at which new digital values are sampled from the analog signal. The rate of new values is called the *sampling rate* or *sampling frequency* of the converter.

A continuously varying band limited signal can be sampled (that is, the signal values at intervals of time T, the sampling time, are measured and stored) and then the original signal can be *exactly* reproduced from the discrete-time values by an interpolation formula. The accuracy is limited by quantization error. However, this faithful reproduction is only possible if the sampling rate is higher than twice the highest frequency of the signal. This is essentially what is embodied in the Shannon-Nyquist sampling theorem.

Since a practical ADC cannot make an instantaneous conversion, the input value must necessarily be held constant during the time that the converter performs a conversion (called the*conversion time*). An input circuit called a sample and hold performs this task—in most cases by using a capacitor to store the analog voltage at the input, and using an electronic switch or gate to disconnect the capacitor from the input. Many ADC integrated circuits include the sample and hold subsystem internally.

Accuracy

An ADC has several sources of errors. Quantization error and (assuming the ADC is intended to be linear) non-linearity is intrinsic to any analog-to-digital conversion. There is also a so-called *aperture error* which is due to a clock jitter and is revealed when digitizing a time-variant signal (not a constant value).

These errors are measured in a unit called the *LSB*, which is an abbreviation for least significant bit. In the above example of an eight-bit ADC, an error of one LSB is 1/256 of the full signal range, or about 0.4%.

Quantization error

Quantization error is due to the finite resolution of the ADC, and is an unavoidable imperfection in all types of ADC. The magnitude of the quantization error at the sampling instant is between zero and half of one LSB.

In the general case, the original signal is much larger than one LSB. When this happens, the quantization error is not correlated with the signal, and has a uniform distribution. Its RMS value is the standard deviation of this distribution, given by $\frac{1}{\sqrt{12}}\mathrm{LSB} \approx 0.289\,\mathrm{LSB}$. In the eight-bit ADC example, this represents 0.113% of the full signal range.

At lower levels the quantizing error becomes dependent of the input signal, resulting in distortion. This distortion is created after the anti-aliasing filter, and if these distortions are above 1/2 the sample rate they will alias back into the audio band. In order to make the quantizing error independent of the input signal, noise with amplitude of 1 quantization step is added to the signal. This slightly reduces signal to noise ratio, but completely eliminates the distortion. It is known as dither.

Non-linearity

All ADCs suffer from non-linearity errors caused by their physical imperfections, resulting in their output to deviate from a linear function (or some other function, in the case of a deliberately non-linear ADC) of their input. These errors can sometimes be mitigated by calibration, or prevented by testing.

Important parameters for linearity are integral non-linearity (INL) and differential non-linearity (DNL). These non-linearities reduce the dynamic range of the signals that can be digitized by the ADC, also reducing the effective resolution of the ADC.