Designing Analog to Digital Converter

With an increase in demand of communication electronics for example smartphones, digital video broadcasting, there is a need to design circuits which are fast, consumes less power in order to increase battery life and occupy less chip area to decrease the cost of production and make it more compact. An analog to digital converter is a circuit which takes an analog signal at its input and produce digital output (bits). As most of the processing is done in digital domain because it is fast, consumes less power and takes small area compared to its analog counterpart but the real world is analog and there is always a need of such sophisticated data converters to extract digital bits from analog world for digital signal processing. A typical receiver circuit has an antenna to receive electromagnetic waves followed by radio frequency circuits to amplify the incoming signal as signal becomes weak after travelling for long distances. Also channel noise and noise from other sources plays a vital role in weakening the signal. Radio frequency circuit is followed by an analog to digital converter which translates the signal into digital data. Figure1 shows a block diagram of an analog to digital converter.

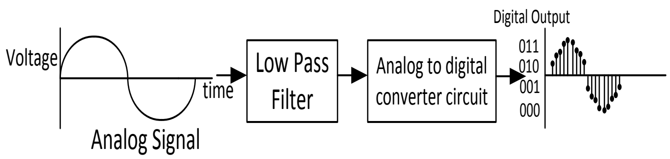

Figure1: Analog to digital converter block diagram

The analog signal is passed through a low pass filter to filter out any high frequency component. Every analog to digital converter do two things 1. It samples the analog input signal 2. Quantize the sampled signal. Sampling is done by applying a clock of specific frequency to the circuit. Depending on the frequency of the clock, signal is sampled. The faster the clock the higher the sampling rate. After sampling, the quantizer circuit quantizes the sampled value by defining a constant amplitude for each sampled data giving unique digital bits for each sample. A low pass filter will prevent aliasing. For a Nyquist-rate converter which satisfies Nyquist criteria i.e. fs (sampling frequency) ≥ 2fmax (maximum frequency of input signal), the signal frequency must be less than ½ of the sampling frequency. If the maximum frequency of the signal is greater than ½ of the sampling frequency, it will alias (shift) into the frequency band of interest and introduce noise into the system which is highly undesirable in any analog to digital converter. That's why there is a need of a low pass filter before analog to digital converter circuit to diminish high frequency components and prevent a shift.

Different types of analog to digital converters are in the market. Flash, pipeline analog to digital converters are some of them. Every analog to digital converter is designed for a certain resolution. A resolution is the number of final digital bits obtained at data converters output. Like 4-bit, 6-bit, 10 bit. Every analog to digital converter is intended for specific applications. Like for applications where a fast conversion is required with high speed, low resolution, where there is no need to worry about power consumption and size, flash analog to digital converter is a good choice where flash here means a sudden burst of digital bits after applying an analog signal at the input. For applications which requires a high speed, medium to high resolution, medium area and less power dissipation, pipeline analog to digital converter is a better choice.

A pipeline analog to digital converter has many stages to resolve the digital data. Number of stages depends on the resolution of the converter. Each stage resolves a portion of the overall digital data. A final digital data is not available until all the intermediate stages resolves its portion of data. After all the data in the pipeline is processed, final digital data is available at the output of the converter. To check the performance of an analog to digital converter if it is working well a real sinusoid signal is applied at its input as the real analog signal is a sine wave and tests are performed to see if the analog to digital converter is performing according to the requirements or not and if there is any high frequency component (distortion) introduced by the analog to digital converter during conversion. A fast Fourier transform is computed on the sampled data at the output of the analog to digital converter. A fast Fourier transform is basically converting the time domain signal into frequency domain. A spectrum is plotted from 0 to fs/2 (sampling frequency) where each spectral component represents a particular frequency. One spectral component represents the input signal which has been applied at the input of analog to digital converter. Other spectral components are the noise from the analog to digital converter. Noise is represented by a noise floor in the spectrum. Above the noise floor are high frequency signals (distortion) which are undesirable except the input signal. These distortion terms are introduced if the analog to digital convertor is not doing fine conversion. The lower the distortion the better. To check the performance of the converter a signal-to-noise and distortion ratio is measured from the spectrum, which is the ratio of the signal power to the noise + distortion power. To verify if the measured signal-to-noise and distortion ratio is OK, a comparison of measured signal-to-noise and distortion ratio is required with the theoretical calculated signal-to-noise and distortion ratio according to equation 1.

$$signal - to - noise\ and\ distortion\ ratio = 6.02n + 1.76dB \qquad (1)$$

Where 'n' is the resolution of the converter. For example, 6-bit analog to digital converter signal-to-noise and distortion ratio should be close to 38dB to determine analog to digital converter performance. Where dB(decibel) is a logarithmic unit which represents the ratio of two power quantities. In this case it is the ratio of signal power which is the analog input and noise + distortion power which is the noise and distortion introduced by the analog to digital converter.