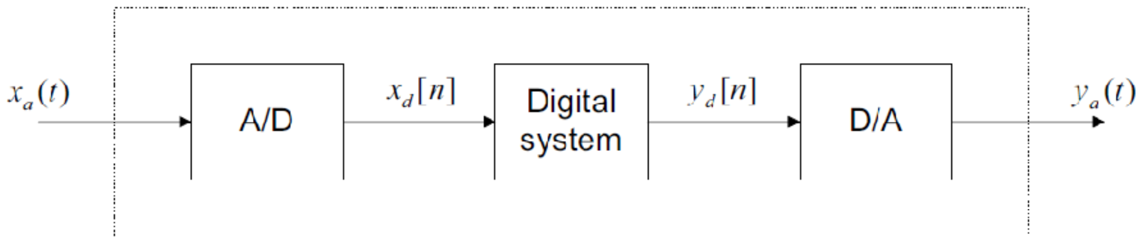
BASIC COMPONENTS OF A DSP SYSTEM

Generic structure:

• In its most general form, a DSP system will consist of three main components, as illustrated in Figure.
• The analog-to-digital (A/D) converter transforms the analog signal *xa*(*t*) at the system input into a digital signal *xd* [*n*]. An A/D converter can be thought of as consisting of a sampler (creating a discretetime signal), followed by a quantizer (creating discrete levels).
• The digital system performs the desired operations on the digital signal *xd*[*n*] and produces a corresponding output *yd* [*n*] also in digital form.
• The digital-to-analog (D/A) converter transforms the digital output *yd*[*n*] into an analog signal *ya*(*t*) suitable for interfacing with the outside world.
• In some applications, the A/D or D/A converters may not be required; we extend the meaning of DSP systems to include such cases.



Discrete-time signals are typically written as a function of an index *n* (for example, *x*(*n*) or x_n may represent a discretisation of *x*(*t*) sampled every *T* seconds). In contrast to Continuous signal systems, where the behaviour of a system is often described by a set of linear differential equations, discrete-time systems are described in terms of difference equations. Most Monte Carlo simulations utilize a discrete-timing method, either because the system cannot be efficiently represented by a set of equations, or because no such set of equations exists. Transform-domain analysis of discrete-time systems often makes use of the Z transform.

Discrete time processing of continuous time signals:

Even though this course is primarily about the discrete time signal processing, most signals we encounter in daily life are continuous in time such as speech, music and images. Increasingly discrete-time signals processing algorithms are being used to process such signals. For processing by digital systems, the discrete time signals are represented in digital form with each discrete time sample as binary word. Therefore we need the analog to digital and digital to analog interface circuits to convert the continuous time signals into discrete time digital form and vice versa. As a result it is necessary to develop the relations between continuous time and discrete time representations.

1. Sampling of continuous time signals:

Let $\{x_c(t)\}$ be a continuous time signal that is sampled uniformly at $t = nT$ generating the sequence $\{x[n]\}$ where

$$x[n] = x_c(nT), \qquad -\infty < n < \infty, \qquad T > 0$$

T is called sampling period, the reciprocal of T is called the sampling fre-quency $f_s = 1/T$. The frequency domain representation of $\{x_c(t)\}$ is given by its Fourier transform.

where the frequency-domain representation of $\{x[n]\}$ is given by its discrete time fourier transform.

To establish relationship between the two representation, we use impulse train sampling. This should be understood as mathematically convenient method for understanding sampling. Actual circuits can not produce contin-uous time impulses. A periodic impulse train is given by:

$$x_p(t) = x_c(t)p(t)$$

using sampling property of the impulse $f(t)\delta(t - t_0) = f(t_0)\delta(t - t_0)$, we get

From multiplication property, we know that:

$$X_p(j\Omega) = \frac{1}{2\pi}[X_c(j\Omega) * P(j\Omega)]$$

The Fourier transform of a impulse train is given by

Where $\Omega_s = 2T$

Using the property that $X(j\Omega)\ \delta(\Omega - \Omega_0) = X(j(\Omega - \Omega_0))$ it follows tha

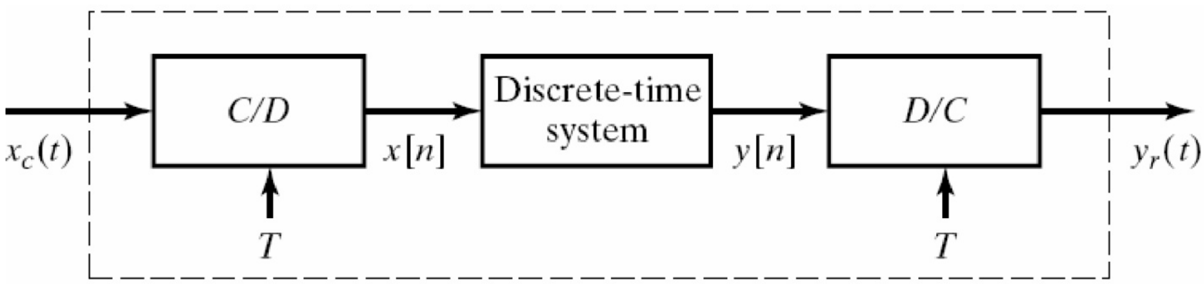
$$X_p(j\Omega) = 1/t\, X_c(j\Omega - k\Omega_s)$$

Thus $X_p(j\Omega)$ is a periodic function of Ω with period Ω_s, consisting of super-position of shifted replicas of

$X_c(j\Omega)$ scaled by 1/T . Figure 8.3 illustrates this for two cases.

If $\Omega_m < (\Omega_s - \Omega_m)$ or equivalently $\Omega_s > 2\Omega_m$ there is no overlap between shifted replicas of $X_c(j\Omega)$, whereas with $\Omega_s < 2\Omega_m$, there is overlap. Thus if $\Omega_s > 2\Omega_m$, $X_c(j\Omega)$ is faithfully replicated in $X_p(j\Omega)$ and can be recovered from $x_p(t)$ by means of lowpass filtering with gain T and cut of frequency between Ω_m and $\Omega_s\Omega_m$. This result is known as Nyquist sampling theorem.

A major application of discrete-time systems is in the processing of continuous-time signals.



The overall system is equivalent to a continuous-time system, since it transforms the continuous-time input signal $x_s(t)$ into the continuous time signal $y_r(t)$.

Sampling Theorem: Let xc(t) be a bandlimited signal with Xc(jΩ) = 0, for |Ω| > Ωm. Then Xc(t) is uniquely determined by its samples x[n] = xc(nT), −∞ < n < ∞, if

$$\Omega_s - \frac{2\pi}{T} > 2\Omega_m$$

The frequency 2Ωm is called Nyquist rate, while the frequency Ωm is called the Nyquist frequency.

The signal xc(t) can be reconstructed by passing xp(t) through a lowpass filter.

The effect of underselling: Aliasing

We have seen earlier that spectrum Xc(jΩ) is not faithfully copied when Ωs < 2Ωm. The terms in overlap. The signal xc(t) is no longer recoverable

From xp(t). This effect, in which individual terms in equation overlap is called aliasing.

For the ideal low pass signal

$$h_r(nT) = \begin{cases} 1 & n = 0 \\ 0 & n \neq 0 \end{cases}$$

Hence xr(nT) = xc(nT), n = 0,±1,±2.......
Thus at the sampling instants the signal values of the original and reconstructed Signals are same for any sampling frequency.

Source : http://msk1986.files.wordpress.com/2013/09/dsp-unit-1.pdf