BASEBAND SAMPLING

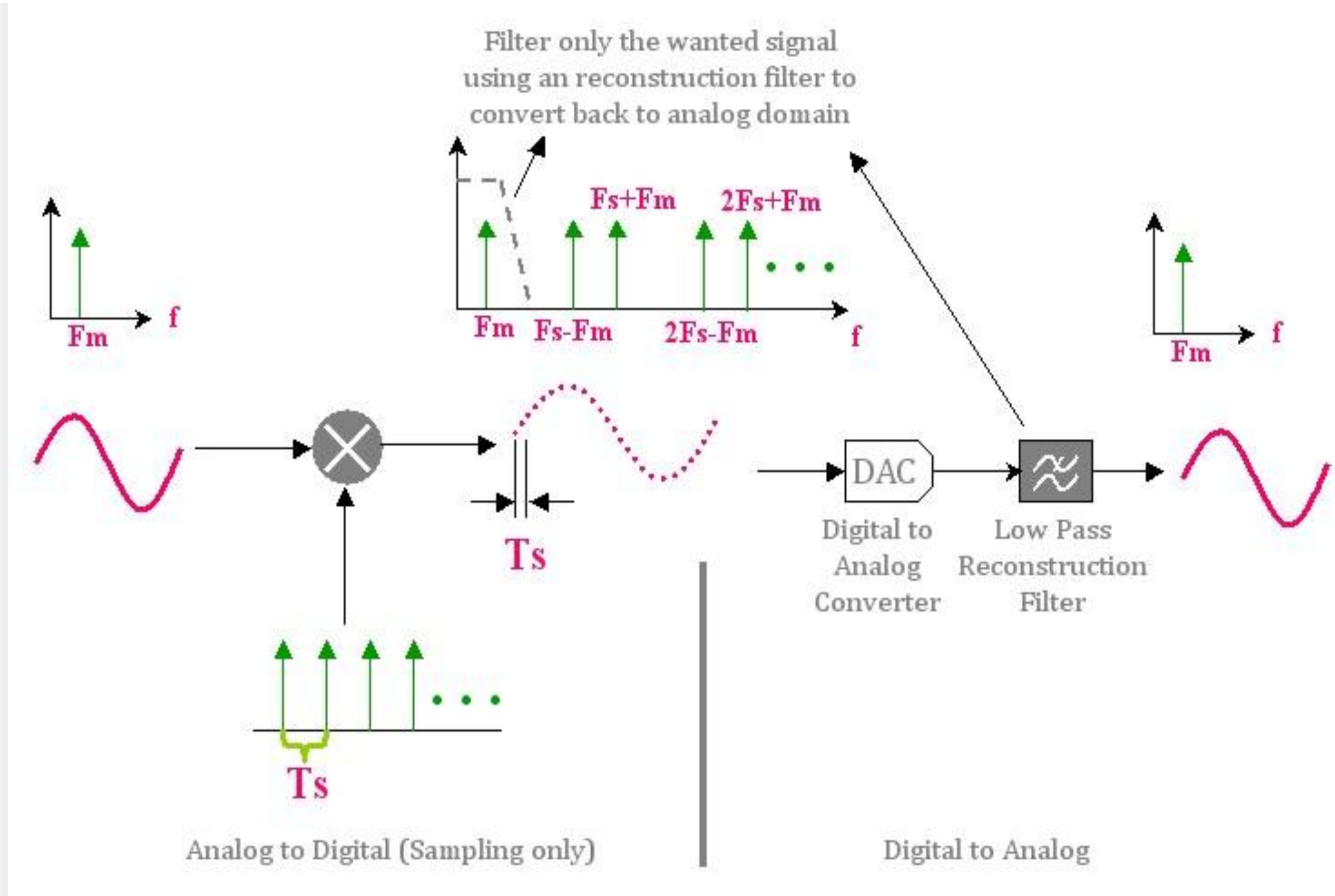
If the signal is confined to a maximum frequency of Fm Hz, in other words, the signal is a baseband signal (extending from 0 Hz to maximum Fm Hz).

In order for a faithful reproduction and reconstruction of an analog signal that is confined to a maximum frequency Fm, the signal should be sampled at a Sampling frequency (Fs) that is greater than or equal to twice the maximum frequency of the signal.

$$F_s \geq 2F_m$$

Consider a 10Hz sine wave in analog domain. The maximum frequency present in this signal is Fm=10 Hz (obviously no doubt about it !!!). Now, to satisfy the sampling theorem that is stated above and to have a faithful representation of the signal in digital domain, the sampling frequency can be chosen as Fs >=20Hz. That is, we are free to choose any number above 20 Hz. Higher the sampling frequency higher is the accuracy of representation of the signal. Higher sampling frequency also implies more samples, which implies more storage space or more memory requirements. In time domain, the process of sampling can be viewed as multiplying the signal with a series of pulses (“pulse train) at regular interval – Ts.

In frequency domain, the output of the sampling process gives the following components – Fm (original frequency content of the signal), Fs±Fm,2Fs±Fm,3Fs±Fm,4Fs±Fm and so on and on…



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Now the sampled signal contains lots of unwanted frequency components (Fs±Fm,2Fs±Fm,…). If we want to convert the sampled signal back to analog domain, all we need to do is to filter out those unwanted frequency components by using a “reconstruction” filter (In this case it is a low pass filter) that is designed to select only those frequency components that are upto Fm Hz. The above process mentions only the sampling part which samples the incoming analog signal at regular intervals. Actually a quantizer will follow the sampler which will discretize (“quantize”) amplitude levels of the sampled signal. The quantized amplitude levels are sent to an encoder that converts the discrete amplitude levels to binary representation (binary data). So when converting the binary data back to analog domain, we need a Digital to Analog Converter (DAC) that converts the binary data to analog signal. Now the converted signal after the DAC contains the same unwanted frequencies as well as the wanted component. Thus a reconstruction filter with proper cut-off frequency has to placed after the DAC to filter out only the wanted components.

Source : http://www.gaussianwaves.com/2011/07/sampling-theorem-baseband-sampling/