# Digital Signal Processing Oversampled Analog to Digital Conversion with Direct Quantization

D. Richard Brown III

# Oversampled ADC System Model



Assumptions:

- The input signal is band limited so that  $X_a(j\Omega) = 0$  for all  $|\Omega| \ge \Omega_N$ .
- ► The input signal is stationary with autocorrelation  $\phi_{x_a x_a}(\tau)$  and power spectrum  $\Phi_{x_a x_a}(j\Omega)$ .

• The C/D block oversampled by a factor of M, i.e.,  $\Omega_s = 2M\Omega_N$ . We have seen previously that, when M = 1, the SNR (in dB) of a B + 1 bit uniform quantizer with step size  $\Delta = \frac{X_m}{2^B}$  is

$$\operatorname{SNR}_Q = 10 \log_{10} \left( \frac{\sigma_x^2}{\sigma_e^2} \right) = 6.02B + 10.8 - 20 \log_{10} \left( \frac{X_m}{\sigma_x} \right)$$

Question: What happens when M > 1?

#### Quantization Error Noise Analysis



Under the usual white-noise statistical assumptions about the quantization error, we have  $\Phi_{ee}(e^{j\omega}) = \sigma_e^2$ . Hence the quantization noise propagates as



and the power of the quantization noise at the output of the downsampler can be computed as

$$\sigma_{x_{de}}^2 = \frac{1}{2\pi} \int_{-\pi}^{\pi} \frac{\sigma_e^2}{M} d\omega = \frac{\sigma_e^2}{M}.$$

DSP: Oversampled ADC with Direct Quantization

### Analysis of Signal Part (part 1 of 2)



We can analyze the signal component similarly:



Note that the signal power is unchanged through the LPF and downsampler.

# Analysis of Signal Part (part 2 of 2)

Note that the power of the CT input signal is

$$\sigma_{x_a}^2 = \frac{1}{2\pi} \int_{-\Omega_N}^{\Omega_N} \Phi_{x_a x_a}(j\Omega) \, d\Omega$$

The power of the sampled DT signal is

$$\sigma_x^2 = \frac{1}{2\pi} \int_{-\pi}^{\pi} \Phi_{xx}(e^{j\omega}) \, d\omega$$

but since

$$\Phi_{xx}(e^{jw}) = \frac{1}{T} \sum_{k=-\infty}^{\infty} \Phi_{x_a x_a} \left( j\frac{\omega}{T} - \frac{j2\pi k}{T} \right)$$

we have

$$\sigma_x^2 = \frac{1}{2\pi} \int_{-\pi}^{\pi} \frac{1}{T} \sum_{k=-\infty}^{\infty} \Phi_{x_a x_a} \left( j \frac{\omega}{T} - \frac{j 2\pi k}{T} \right) \, d\omega = \frac{1}{2\pi} \int_{-\pi/M}^{\pi/M} \frac{1}{T} \Phi_{x_a x_a} \left( j \frac{\omega}{T} \right) \, d\omega.$$

We can substitute  $\Omega=\frac{\omega}{T}$  and note that  $d\omega=Td\Omega$  to write

$$\sigma_x^2 = \frac{1}{2\pi} \int_{-\Omega_N}^{\Omega_N} \Phi_{x_a x_a} \left( j\Omega \right) \, d\Omega = \sigma_{x_a}^2.$$

Hence the power of the analog input and the discrete-time signal are identical.

#### Oversampled ADC With Direct Quantization SNR Analysis

We can compute the SNR (in dB) of a B+1 bit uniform quantizer with step size  $\Delta=\frac{X_m}{2^B}$  and oversampling factor M as

$$SNR_Q = 10 \log_{10} \left( \frac{\sigma_{x_{da}}^2}{\sigma_{x_{de}}^2} \right)$$
$$= 10 \log_{10} \left( \frac{\sigma_x^2}{\sigma_e^2/M} \right)$$
$$= 10 \log_{10} \left( \frac{12 \cdot 2^{2B} \sigma_x^2 M}{X_m^2} \right)$$
$$= 6.02B + 10.8 - 20 \log_{10} \left( \frac{X_m}{\sigma_x} \right) + 10 \log_{10}(M)$$

Remarks:

- ▶ Main idea: Oversampling by *M* doesn't affect the signal power but reduces the noise power by a factor of *M*.
- Doubling M and keeping B fixed gives a 3 dB SNR improvement.
- Quadrupling M and using one less quantizer bit gives the same SNR.