Lecture 2 Data Conversion

Objectives:

■Review signal conversion in context of DSP systems

■Important issues relating to signal conversion including:

–Sampling and aliasing

–Signal to quantization noise ratio

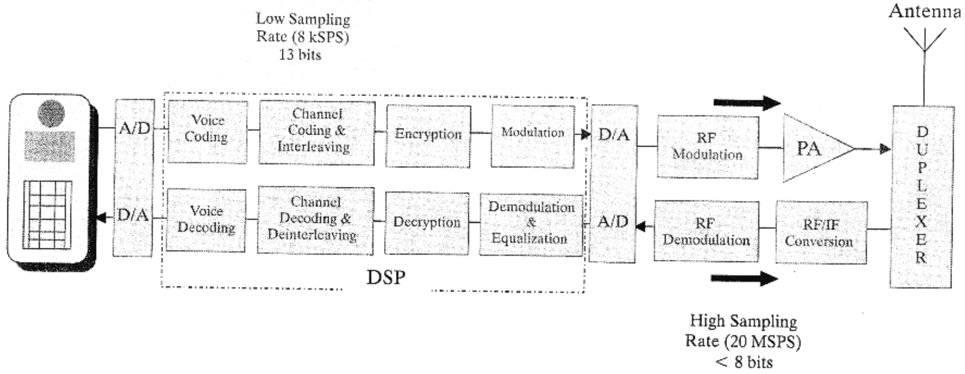
–Harmonic distortion

–Sampling clock jitter

–Oversampling converters

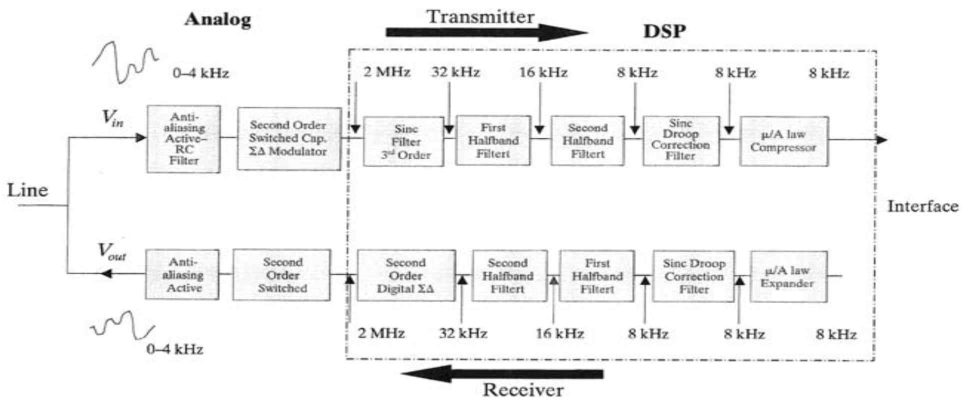
Reference: "DSP System Design using the TMS320C6000" by N. Kehtarnavaz & M. Keramat, Prentice Hall 2001

Typical wireless DSP system



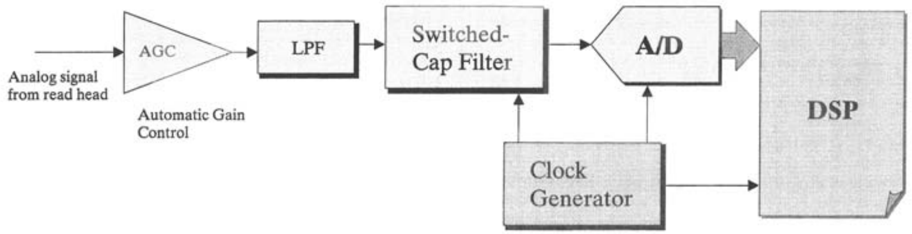
- Two data converters required:
  - low speed (8 kSPS) and high resolution (13bits) for voice
  - high speed (20 MSPS) and low resolution (8-bits) for RF
- DSP functions integrated - mixture of ASIC & DSP processors

Typical PCM voiceband DSP system



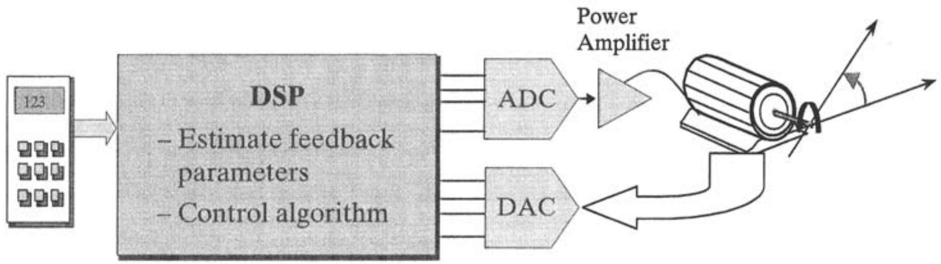
- Voiceband processing often use a low cost converter based around a sigma-delta ADC/DAC system with a reasonable amount of DSP hardware.

Typical hard disk DSP system



- Hard disk interface electronics also rely heavily on digital signal processing. Data conversion is normally done at very high frequency (200 MSPS or more) and low resolution (6 bits).
- Complex feedback system between DSP, ADC and analogue circuits.
- Very demanding example: high performance, low power and low cost!

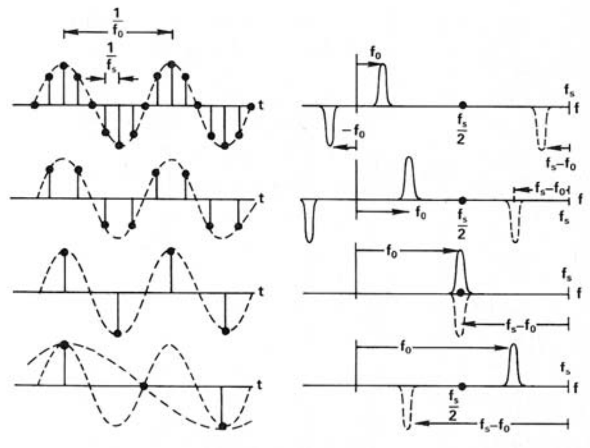
Typical motor control DSP system



- DSP processors are often used in control applications.
- Motor control is an example - low speed and moderate resolution.
- Data conversion requirements tend to be modest. Control algorithm can be very complex.

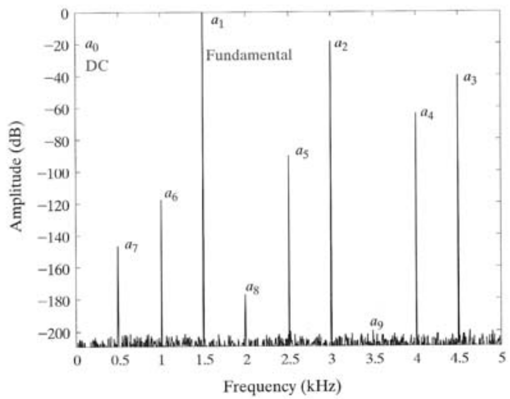
Aliasing in Sampling

- Sampling theorem: given a signal which is bandlimited to f_{max} sampling frequency fs must be at least $2*f_{max}$.
- $2*f_{max}$ is called the **Nyquist** frequency.
- Sampling the signal below this rate will cause signal corruption - **aliasing**



Harmonic Distortion and sampling

- Real life circuits are not perfectly linear
- Output of linear (analogue) circuit may have distortion
- Subsequent sampling (e.g. saturation of the input stage of a ADC system) will cause spurious signal appearing in the baseband

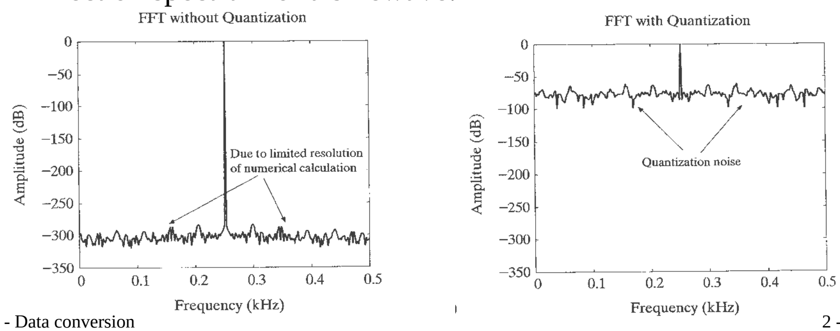


Quantization effect of A/D converters

- The minimum step that an A/D can resolve is: $1\ LSB = \Delta = \frac{V_{ref}}{2^N}$
- Quantization noise n_q is assumed to be signal independent and is uniformly distributed over -0.5 LSB and 0.5 LSB, then the quantization noise variance is:

$$\sigma_q^2 = \mathrm{E}\left[n_q^2\right] = \frac{1}{\Delta}\int_{-\Delta/2}^{\Delta/2} n_q^2\, dn_q = \frac{\Delta^2}{12}$$

- Effect on spectrum of a sinewave:



Signal-to-Noise Ratio of A/D converter

- For a sinusoidal signal with an amplitude of A_m:

$$10\log\frac{P_s}{P_n} = 10\log\frac{(A_m)^2/2}{\frac{1}{12}\left(V_{ref}/2^N\right)^2}$$

$$SNR_{max} = 10\log\frac{\left(V_{ref}/2\right)^2/2}{\frac{1}{12}\left(V_{ref}/2^N\right)^2} = 10\log\frac{3}{2}2^{2N} = 6.02N + 1.76\,dB$$

Signal-to-Noise Ratio for Gaussian signal

- Assume a Gaussian signal with zero mean and standard deviation σ_s such that: $V_{ref} = K\sigma_s$

$$SNR_{max} = 10\log\frac{\sigma_s^2}{\sigma_q^2} = 6N + 10.8 - 20\log_{10} K\ dB$$

- For a linear time-invariant system (e.g. FIR, IIR filters), the noise variance at the output caused by input quantization is:

$$\sigma_o^2 = \sigma_q^2 \sum_n h^2[n]$$

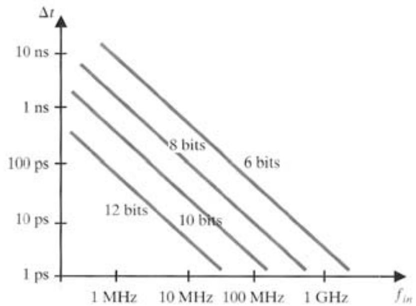
where h[n] = impulse response of the system

Effect of sampling time jitter

- Jitters associated with sampling clock contribute to additive noise
- For sinusoidal signals, maximum allowable time jitter which results in less than ½LSB is given by:

$$\Delta t_{max} = \frac{1}{\pi f_{in} 2^{N+1}}$$

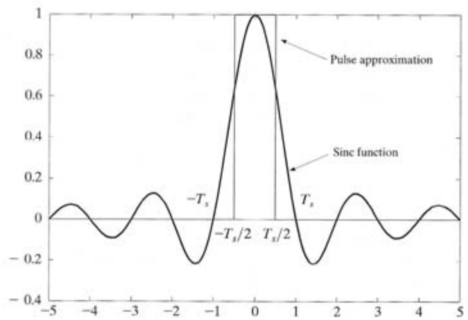
- The following graph provides a useful guideline:



Signal reconstruction with D/A converter

- Perfect reconstruction can be achieved by filtering the sampled signal with a brickwall filter, which is the same as convoluting the sampled signal with a ***sinc*** function:

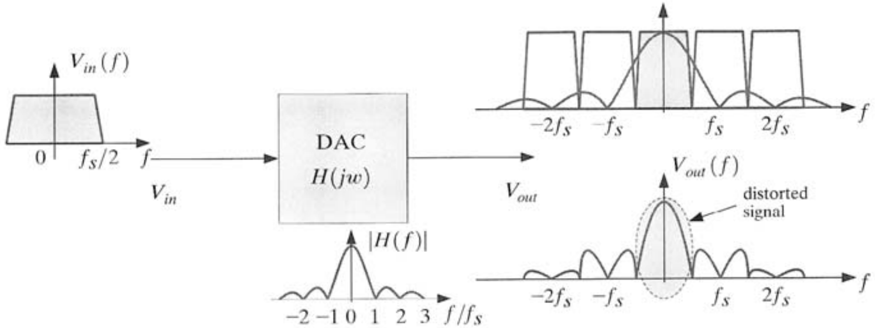
$$v_a(t) = \sum_{k=-\infty}^{\infty} v_a(kT_s)\left[\operatorname{sinc}\left(\frac{t-kT_s}{T_s}\right)\right]$$



Zero-order hold effect of D/A converter

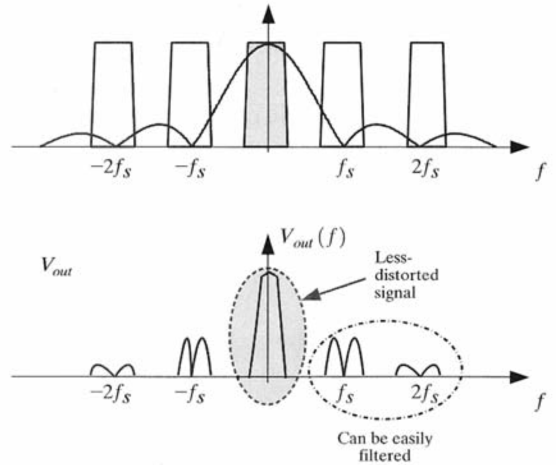
- This is difficult to achieve, therefore use approximation of sample-and-hold function with a D/A converter. The transfer function of a D/A converter is:

$$H(j\omega)=\frac{1}{j\omega}-\frac{1}{j\omega}e^{-j\omega T_s}=\frac{\sin(\omega T_s/2)}{\omega T_s/2}e^{-j\omega T_s/2}=\text{sinc}\left(\frac{f}{f_s}\right)e^{-j\pi f/f_s}$$



How to reduce this D/A error?

- Two approaches to reduce this error:
  - sinc correction: pre-distort signal to compensate for this error
  - over-sampling



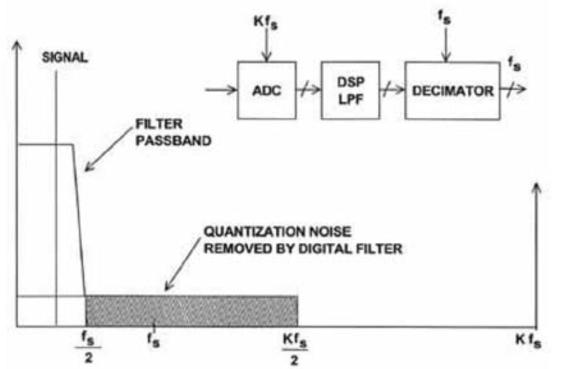
Sigma-Delta A/D converters: why?

- Two types of A/D converters:
  - Nyquist rate: flash A/D, successive approximation A/D
  - Oversampling: sigma-delta
- Advantages of sigma-delta converters:
  - Inherently linear
  - High resolution (16-24 bits)
  - Good for mixed-signal IC processes (e.g. CMOS)
  - No sample-and-hold circuit required
- Disadvantages:
  - Limited to voiceband and audio
  - Difficult to multiplex one ADC to multiple channels
- Almost all audio ADCs use sigma-delta technqiues

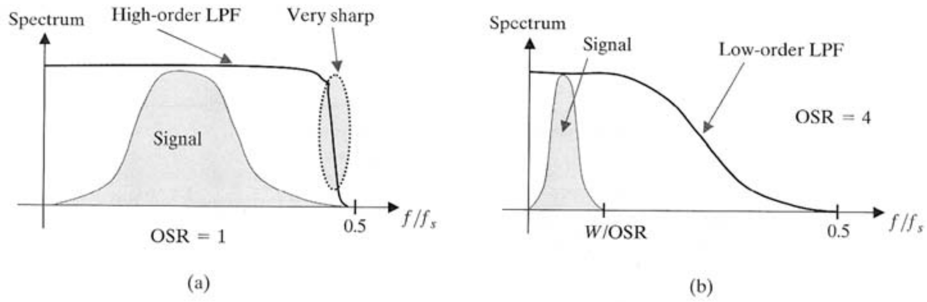
Oversampling A/D

- If the oversampling ratio OSR is: $OSR=\frac{f_s}{2f_{\max}}$
- SNR can be improved according to:

$$SNR_{oversampling}=6.02N+1.76+10\log(OSR)\ dB$$

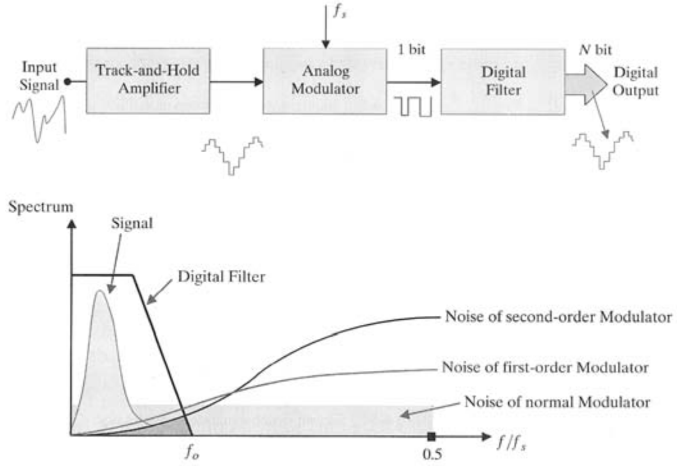


Easy antialiasing filter



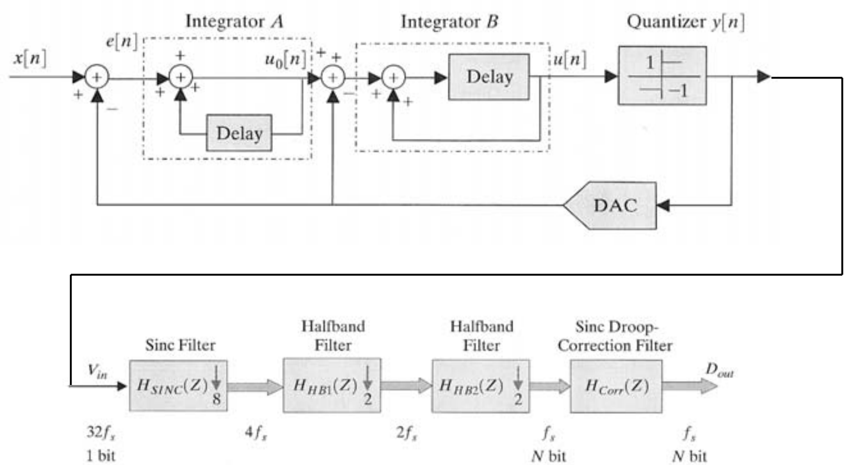
- Oversampling A/D requires simple lowpass filter at input

Sigma-delta modulator for noise-shaping

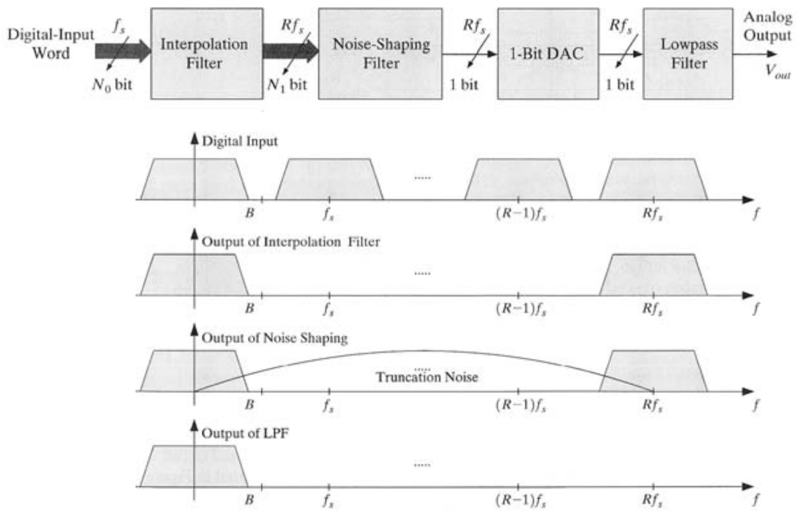


- Use sigma-delta modulator to push the quantization noise power towards high frequency end, which is then filtered out

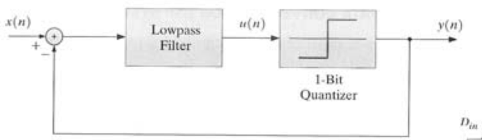
Block diagram of a sigma-delta converter



Oversampled D/A converter

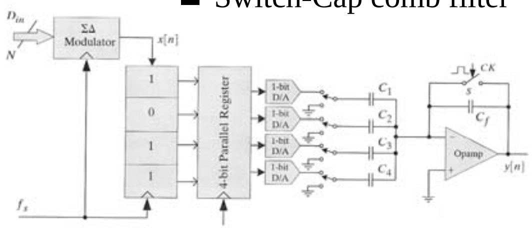


Oversampled D/A converter

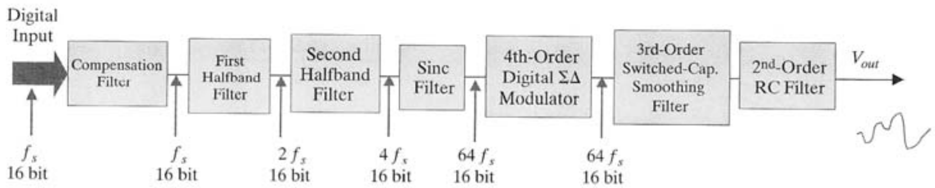


- 1st order ΣΔ modulator

- Switch-Cap comb filter



- Typical 16-bit ΣΔ DAC



TLC320AD535 sigma-delta codec