

DSP Notes: Digital vs. Analog Signal Processing

Professor Fred DePiero, CalPoly State University

Why DSP? Why Not Analog Signal Processing?

Advantages of Digital Processing

- + Linear and nonlinear math operations work over a wide dynamic range of signal, 2^{31} to 2^{-31} for standard floating point. Also a suite of operations, like $\cos()$, $\text{atan}()$, $\text{sqrt}()$, $\log()$ are available.
- + Higher order filters can be implemented with a relatively low incremental cost. Additional memory and computations only.
- + Filter design techniques provide a relatively high degree of freedom in spectral shaping, as in the Frequency Sampling method, for example.
- + No tuning of analog components (R,L,C) during production or during maintenance.
- + Good version control. Burn filter coefficients into memory and these will never change from one unit to the next.
- + Software-based implementations require no custom hardware - just use standard signal I/O boards and write custom software.
- + Small and rugged implementation using mixed-type VLSI, combining both DSP and analog I/O on a single chip.
- + Adaptive filters become practical.
- + Data compression becomes practical.

Advantages of Analog Processing

- + No computer, no A/D or D/A.
- + Better for high frequency applications, when time available for data conversion (A/D) or for computations is too limited.
- + Simpler implementation as far as an overall component count.
- + Less expensive overall, provided no computer is available. (Possibly more expensive if a computer is already available.)
- + No counterpart to the problem of limit cycles that occur in Digital IIR filters. Limit cycles are a never-ending low amplitude oscillation in the output value of a filter which is caused by rounding effects introduced by finite precision math operations.
- + Smoother transition of low-level analog signals into the noise floor, as opposed to the output of a (dithering) D/A which generates a low duty cycle square wave (peak voltage corresponding to a count of 1) when outputting a digital value between 0 and 1.

+ No need to impose certain standards. For example specific parameters associated with the digitization process, such as quantization fineness and sampling rate. Once these standards are set they are difficult to change or improve. A current debate over standards is the sufficiency of 16 bits / 44.1 kHz sampling for CDs.

Comparisons of Analog / Digital Processing

- Cost of high Q for digital filter: high precision coefficients and computations.
- Cost of high Q for analog filter: high precision components.

- SNR of a digital signal determined by the maximum discrete amplitude and the resolution of the A/D (volts/count).
- SNR of analog signal determined by power supply rails and by noise floor.

- Dynamic spectral shaping in a digital filter via on-line coefficient changes.
- Dynamic spectral shaping in an analog filter via various means: (1) Switched capacitor circuits can change cutoff frequencies, (2) Q of a bandpass filter can be changed by varying gains.

Properties of a Practical, Near-Ideal Digital Filter

- Must be causal, with finite length.
- Some ripple will occur in frequency response due to Gibbs phenomena.
- Will have finite width transition region, also due to Gibbs.
- Complex poles and zeros must occur in pairs, to ensure real coefficients.
- Poles must remain inside the unit circle for stability.
- Zeros do not effect stability, regardless of location in plane.
- Linear phase characteristic of $\angle H(\omega)$ corresponds to a filter delay that is constant for all frequencies.
- For a causal, real-valued filter the magnitude and phase of $H(\omega)$ can not be independently specified.