SUMMARY

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Nowadays digital audio is ubiquitous, it can be easily acquired and reproduced with small and portable systems. The human hearing system evolved to an high accuracy in both time and frequency domains. To match this feature, a great effort is needed to design the device that converts the digital signal to its analog counterpart for physical fruition by a human listener. This system is called Digital-to-Analog Converter (DAC) and can be realized in many ways. Each implementation presents a slightly different sound signature and, for High Fidelity audio reproduction, this component must be able to accurately resolve the original analog signal. It can be difficult and expensive, in particular for people looking for the ultimate sound reproduction experience. Many stand-alone high-end DACs are available on the market but there is no clear winner, also due to the subjectivity of the end-user auditory tastes. It is thus important to be able to create a converter as sonically transparent as possible to let also the pickiest consumer enjoy the music content.

There are various issues associated with DAC designs. The main source of error is the analog part of the system, as the digital part exploits its intrinsic mathematical abstraction layer to reduce the error associated to its operations to the minimum. The analog part is prone to static and dynamic mismatch errors and susceptible to various noise contaminations. Fortunately, it is possible to employ clever Digital Signal Processing (DSP) algorithms to reduce the impact of mismatch and, to some extent, of noise on the analog stage. Modern Complementary Metal–Oxide–Semiconductor (CMOS) digital circuits can correct for many analog non-idealities with a relatively small resource utilization.

The most powerful idea in this field is to oversample the input signal and apply spectral shaping to error sources in order to force the error contribution to occupy a spectral zone where it can be successfully filtered out without affecting the original input signal. The quantization error, the static mismatch error, or the dynamic Inter-Symbolic Interference (ISI) error can be all shaped resorting to appropriate algorithms. This paves the way for very compact and high performance integrated circuits realizations thanks to modern deep sub-micron CMOS technological processes.

Ultimate performance has not been reached yet, for both high-end stand-alone systems and mostly-digital integrated circuits. This thesis aims to propose some novel ways to reach the state of the art for modern audio DACs.

Oversampling usually requires an high quality interpolator to increase the sample rate. The first part of the novel contributions section revolves around two new ways to perform high quality interpolation. The first one optimizes FIR filtering by adding a recursion scheme, which is particularly efficient for linear phase filter kernels that can be represented sparsely in the frequency domain. The second method works directly in the frequency domain in a block-wise operation using a novel mix of Discrete Sine Transform (DST) and Discrete Cosine Transform (DCT) to reduce border artifacts.

After that, some new modulation schemes are explored. The first one embeds the analog mismatch error shaping directly inside the main noise shaper. This extends the error correction scheme effectiveness while resorting to the classical Delta Sigma framework, which simplifies the system design. The second one works on the system stability, proposing a simple yet effective way to ensure a stable operation with a very little computational overhead. Next, a way to increase time-domain and frequency-domain signal fidelity is presented. The time-interleaving of two signals is used to compensate errors, leading to a simplified look-ahead scheme. Lastly, a mostly-digital low-power system is presented. It exploits the combination of a low-oversampling multi-level Delta Sigma modulator, Dyadic Digital Pulse Modulation (DDPM) and a shift-register-like DAC output to create a multi-level analog signal stemming from a two-level digital bitstream.

All the presented techniques are orthogonal in the solution space. They can potentially combined together to form an interesting DAC with a high quality interpolation scheme, intrinsic static mismatch error shaping, unconditional stability and high Signal-to-Noise Ratio (SNR).