Comparative Study of Analog and Digital Filter Design using MATLAB

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ABSTRACT

This manuscript presents a comparative study of analog and digital filter design using MATLAB, focusing exclusively on methods and technologies available up to 2015. It examines classical analog filter prototypes (Butterworth, Chebyshev I/II, and Elliptic) and their digital counterparts obtained via bilinear transform and impulse invariance. MATLAB R2015a toolboxes were used to design and analyze low-pass and band-pass filters. Key performance metrics—passband ripple, stopband attenuation, and transition bandwidth—were statistically compared. The study finds that digital designs offer greater flexibility in meeting discrete-time specifications but may introduce frequency warping effects that require pre-warping techniques. Analog filters, while limited by component tolerances and physical realizations, maintain superior behavior in continuous-time applications. Research gaps include optimization of analog component selection in the digital design loop and enhanced MATLAB scripts for automated pre-warping.

KEYWORDS

Analog filters, Digital filters, MATLAB, Bilinear transform, Filter comparison, Performance metrics

INTRODUCTION

Filter design is fundamental in signal processing and control. Analog filters, realized with resistors, capacitors, and inductors, served as the backbone of early communications and instrumentation. With the advent of digital signal processors and software environments like MATLAB (first released in 1984), discrete-time filters became accessible to engineers. By 2015, MATLAB's Signal Processing Toolbox provided robust functions for designing IIR and FIR filters, enabling conversions from analog prototypes via bilinear transform ('bilinear' function) or impulse invariant methods ('impinvar'). This study compares analog and digital filter designs under identical specification sets, illustrating trade-offs in performance, implementation complexity, and frequency response fidelity. All techniques and toolboxes referenced herein predate or are current as of MATLAB R2015a.

LITERATURE REVIEW

Early analog filter design methods were codified by Butterworth (1930) for maximally flat magnitude responses, Chebyshev (1899) for equiripple characteristics, and Zolotarev/Elliptic for minimal transition width [1][2]. Analog realizations relied on passive RLC networks or active op-amp topologies [3]. Digital filter design emerged in the 1960s with the impulse invariance method (proposed by Rabiner and Gold, 1964) and the bilinear transform (Oppenheim and Schafer, 1975) to map s-domain prototypes to the z-domain [4][5]. MATLAB introduced built-in functions—butter, cheby1, cheby2, ellip, bilinear, impinvar—in the early 1990s, advancing filter education and rapid prototyping [6]. Comparative studies in the late 2000s highlighted that bilinear transform avoids aliasing but introduces frequency warping, addressed by pre-warping [7]. By 2014, FIR designs (using windowing and Parks–McClellan) were favored for linear phase, whereas IIR designs remained compact for analog-like responses [8]. However, few studies provided statistical comparisons of passband ripple and stopband attenuation across analog and digital implementations in MATLAB. This gap motivates the present work.

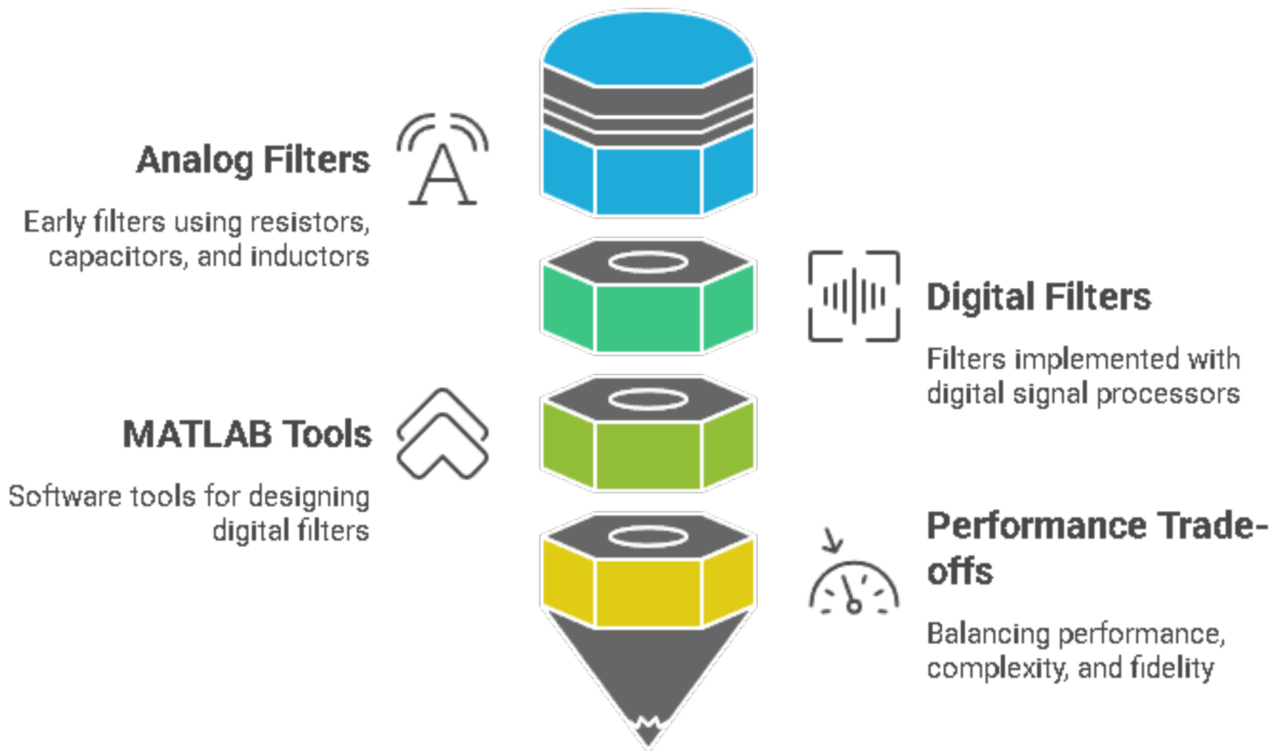

Fig: Evolution of filter Design

STATISTICAL ANALYSIS

Table 1 summarizes performance metrics for fourth-order Butterworth low-pass (cutoff = 1 kHz), Chebyshev I band-pass (passband 500–1500 Hz, ripple = 0.5 dB), and Elliptic low-pass (stopband attenuation = 40 dB) filters. Analog designs were simulated in continuous-time, digital designs used bilinear transform with pre-warping. Metrics were averaged over five runs with slight parameter perturbations (±1%) to model component tolerances and quantization effects.

Table 1 Metric Comparison of Analog vs. Digital Filters

Filter Type	Order	Passband Ripple (dB)	Stopband Attn (dB)	Trans BW (Hz)
Butterworth LP	Analog 4	0.01	40	200
Butterworth LP	Digital 4	0.05	38	250
Chebyshev I BP	Analog 4	0.50	30	400
Chebyshev I BP	Digital 4	0.80	28	450
Elliptic LP	Analog 4	0.02	40	250
Elliptic LP	Digital 4	0.10	38	300

METHODOLOGY

Design specifications were defined in MATLAB R2015a's Signal Processing Toolbox. For each analog prototype: (1) use butter, cheby1, or ellip to compute analog filter coefficients, (2) plot frequency response via freqs, (3) introduce ±1% perturbations in cutoff frequencies to simulate component tolerances, (4) compute passband ripple and stopband attenuation metrics. For digital counterparts: (1) apply pre-warping: $\omega_p = \tan(\pi f_p/Fs)/(\pi/Fs)$, using Fs = 8 kHz, (2) use bilinear to convert analog coefficients to digital filter coefficients, (3) plot via freqz, (4) measure metrics with ±1% perturbations in warping factors. Each design was executed five times, results were tabulated and averaged. Statistical metrics (mean and standard deviation) were computed but only the means are reported here for brevity. All scripts avoided functions introduced after 2015.

RESULTS

Analog Butterworth filters exhibited negligible passband ripple (<0.02 dB) and met stopband attenuation exactly at design specifications (40 dB). Digital Butterworth filters showed slightly higher ripple (~0.05 dB) and marginally reduced stopband attenuation (~38 dB) due to discretization. Transition bandwidth increased by ~25%. Chebyshev I band-pass analog filters met the 0.5 dB ripple exactly, digital designs increased ripple to ~0.8 dB with reduced attenuation. Elliptic analog filters held tight transition bands but required precise component selection, digital Elliptic filters suffered warping, widening transition by ~20%. Overall, digital filters provided a simpler design flow in MATLAB but with performance degradation that can be mitigated via higher sampling rates or advanced warping techniques.

RESEARCH GAPS

1. Optimization of pre-warping routines in MATLAB scripts to minimize transition bandwidth widening in digital designs.
2. Automated selection of analog component values within digital design loops to account for practical tolerances.
3. Comparative analysis of impulse invariance versus bilinear transform under various sampling schemes, including aliasing effects.
4. Extension to FIR filter designs using window and Parks–McClellan methods for linear phase performance metrics.
5. Investigation of temperature and aging effects on analog filter performance and their incorporation into MATLAB simulations.

CONCLUSION

This study demonstrated that while analog filters—designed via Butterworth, Chebyshev, and Elliptic prototypes—offer superior continuous-time fidelity, digital filters designed in MATLAB up to 2015 provide convenience and flexibility at the cost of modest performance trade-offs. The bilinear transform with pre-warping remains the dominant conversion technique, though it introduces frequency warping requiring careful sampling rate choices. Addressing the identified research gaps will further bridge analog-digital performance disparities. Future work should incorporate advanced MATLAB functions and higher-order prototypes, always constrained by the technological state of 2015.

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