Study of algorithmic approaches to digital signal filtering and the influence of input quantization on output accuracy

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Abstract. This article explores the design and application of intelligent control systems tailored for industrial and production environments, with a particular focus on integrating advanced digital filtering and noise reduction strategies. Traditional adaptive control systems often struggle to maintain flexibility and responsiveness in uncertain and dynamic conditions. In response, the study investigates the potential of intelligent control architectures, enhanced by neural networks and fuzzy logic, to overcome these limitations. A comprehensive comparison is made between classical and modern algorithmic approaches to digital filtering, highlighting the critical role of software-based solutions in suppressing quantization errors and mitigating the influence of random noise in analogue sensor data. The implementation of both recursive and non-recursive digital filter models is addressed through digital signal processing techniques, enabling real-time correction and analysis of incoming signals. Using mathematical tools such as Z-transforms and transfer functions, the paper provides an in-depth performance evaluation of filter behaviour under both deterministic and stochastic inputs. The research supports the broader integration of AI-driven technologies in modern automation systems, paving the way for more adaptive, efficient, and fault-tolerant control mechanisms in complex environments.

1 Introduction

The task of developing and creating predominantly new technical and algorithmic solutions for intelligent control systems for industrial facilities and production systems opens up great prospects for the development of intelligent control technologies

With the use of intelligent technologies in control and management systems, the tasks of replacing traditional adaptive automatic control systems with high-speed systems with high

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flexibility, operating under uncertainty, are successfully solved due to the potential advantage of intelligent control systems.

Implementing circuit-based enhancements to classical adaptive control algorithms using digital technologies while considering the stability requirements of discrete control systems does not always yield the expected outcomes [1,2].

2 Methods and materials

The continued advancement of control systems in technical processes relies on enhancing automatic control systems with intelligent capabilities, leveraging contemporary methods and technologies for processing knowledge. Concurrently, this evolution necessitates the development of a new component base — including neural network architectures and fuzzy logic controllers — designed to facilitate intelligent data processing and advanced control techniques.

Historically, the design of systems for digital filtering relied on different hardware setups, including digital controllers and operational amplifiers tailored to particular frequency characteristics and signal conversion ratios. A digital filter (DF), in essence, is a discrete-time system described by a mathematical equation, which is most commonly executed through software implementation [3,4].

$$y(nT) = -\sum_{m=1}^{M-1} a_m y(nT - mT) + \sum_{k=0}^{N-1} b_k x(nT - kT) \tag{1}$$

Fig. 1. Structural diagram of a digital filter.

The input signal $x(nT)$ and the output response $y(nT)$ of the digital filter (Figure 1) are represented as sequences of quantized values. Nonetheless, during the actual execution of the algorithm, filtering takes place with minor inaccuracies. A key objective of software-algorithmic approaches is to minimize the effects of quantization errors.

In measurement channels, noise suppression becomes a critical evaluative procedure, particularly when working with different analogue sensors installed at specific locations to monitor process parameters and ensure the accuracy and dependability of data transmission. The signals obtained from primary sensors inherently contain various types of noise, which are typically categorized into two main types: steady white noise, characterized by a relatively constant oscillation amplitude, and random pulse-like noise, generally caused by disturbances, the sources of which were discussed in the first chapter of the dissertation. Properly structured noise filtering significantly reduces errors and enhances the fidelity of sensor-derived measurement signals [5,6].

The efficiency of filtering noise and interference is influenced by time, and this performance is evaluated under the condition $t_{\min} \leq T$, where T represents the signal's quantization interval.

The principal tool for implementing algorithmic functionality is software.

$$y(nT) = \sum_{k=0}^{4} b_k x(nT - kT) = b_0 x(nT) + b_1 x\big((n-1)T\big) + b_2 x\big((n-2)T\big) + b_3 x\big((n-3)T\big) + b_4 x\big((n-4)T\big) \tag{2}$$

$$H(z) = \sum_{k=0}^{4} b_k z^{-k} \tag{3}$$

or transfer function:

To calculate the characteristics of the filter and output signals under a deterministic action, we will perform according to the following method.

Let a block diagram (or graph of a discrete system) containing L nodes and P branches be given, and let$x_k(nT)$ −input digital signal with a quantized sequence, which is formed at the output of the adder in the k-th node,

$X_k(z)$-z- sequence conversion$x_k(nT)$, $k = 1,2..., L$. For each node of the graph, a nodal equation can be written that describes the interaction of sequences or their Z-images.

Take, for example, the implementation of a non-recursive digital filter described by the following equation (see Figure 2).

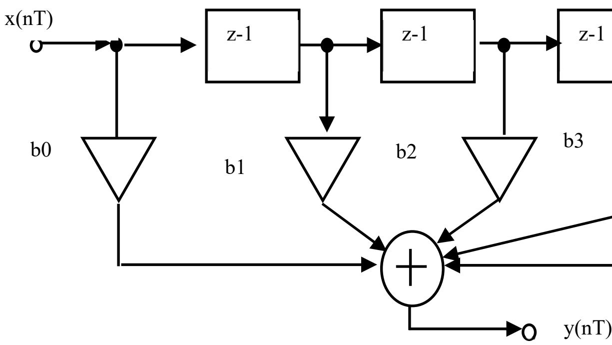

Fig. 2. Recursive digital filter in direct form.

Thus, we obtain a system of L algebraic equations for Z-images$X_k(z)$,$k = 1, ..., L$. By solving this system, one can find the Z-image of the output sequence Y(z) expressed in terms of the Z-image of the input sequence$X(z)$: [2,3]

$$Y(z) = H(z)X(z), \tag{4}$$

from which the transfer function of the system is determined,$H(z)$. The transfer functions of any part of a discrete system are defined similarly.

To estimate the transient duration when a harmonic input signal $x(nT) = e^{j\omega nT}$ is applied to the filter, the output signal $y(nT)$ gradually approaches $n \to \infty$ a steady-state harmonic response at the same frequency ω:

$$y_n(nT) = A(\omega)e^{j\varphi(\omega)}e^{j\omega nT} \tag{5}$$

n this context, H(z) denotes the transfer function of the filter, $A(\omega)\left|H(e^{j\omega T})\right|$ characterizing how the filter responds to different frequency components of the input signal;
$\varphi(\omega) = arg\, H\,(e^{j\omega T})$-PFC of the filter.

It is often necessary to estimate the *real* duration of the transient process, denoted as lT, i.e., to identify the point at which the condition y(nT)≈y_{out}(nT) is approximately satisfied.

Clearly, this condition holds if

$$\sum_{n=0}^{\infty}|h(nT)| \approx \sum_{n=0}^{l}|h(nT)|. \tag{6}$$

The meaning of approximate equality (5) can be refined, for example, as follows: (4) is satisfied if

$$\sum_{n=0}^{\infty}|h(nT)| - \sum_{n=0}^{l}|h(nT)| = \sum_{n=l+1}^{\infty}|h(nT)| < \varepsilon. \tag{7}$$

and

$$\varepsilon \le \varepsilon_1 \sum_{n=0}^{l}|h(nT)|. \tag{8}$$

By assigning a valueε_1—for example, u=0.01u = 0.01u=0.01 to 0.0010.0010.001—and using the known system characteristics $h(nT)$, it is possible to estimate from equation (8) the value of *l* for which the error ε remains within the specified tolerance. The filtering of discrete random sequences is carried out in such a way that a stationary random sequence x$(nT)$is applied to a discrete filter defined by its impulse response $h(nT)$and transfer function:

$$H(z) = \sum_{n=0}^{\infty} h(nT)\, z^{-n} \tag{9}$$

Output Random Filter Sequence$y(nT)$is determined from the relation

$$y(nT) = \sum_{k=0}^{\infty} h(kT)\, x(nT - kT) \quad (10)$$

or from relations in Z-images

$$y(z) = H(z)X(z) \quad (11)$$

Let's find the statistical characteristics of the output sequence$y(nT)$, if the corresponding characteristics of the input sequence are known [6] .

Average powerP_{cpy}output sequence

$$P_{cpy} = P_{cpx} \sum_{n=0}^{\infty} h^2(nT). \quad (12)$$

If the input signal has zero mean$(\mu_x = 0)$, then and

$$\sigma_y^2 = \sigma_x^2 \sum_{n=0}^{\infty} h^2(nT) \quad (13)$$

correlation function$R_y(mT)$output sequence is determined by the average value of the product of functions

$$y(nT) = \sum_{k=-\infty}^{\infty} h(kT)\, x(nT - kT) \quad (14)$$

and

$$y\big((n+m)T\big) = \sum_{k=-\infty}^{\infty} h(lT)\, x(nT + mT - lT) \quad (15)$$

$$R_y(mT) = E[y(nT)y(nT + mT)] = \sum_{k=-\infty}^{\infty} \sum_{l=-\infty}^{\infty} h(kT)h(lT)E[x(nT - kT)x(nT + mT - lT)] \quad (16)$$

Assuming l=k+p, we write

$$R_y(mT) = \sum_{k=-\infty}^{\infty} \sum_{p=-\infty}^{\infty} h(kT)h(kT + pT)E[x(nT - kT)x(nT + mT - kT - pT)]. \quad (17)$$

Note that

$$E[x(nT - kT)x(nT - kT + mT - pT)] = R_x(mT - pT),$$

and denoting

$$\sum_{k=-\infty}^{\infty} h(kT)h(kT + pT) = g(pT) \quad (18)$$

find the connection between the correlation functions of the output and input sequences in the form of a convolution: [2, 3, 7].

$$R_y(mT) = \sum_{p=-\infty}^{\infty} g(pT)\, R_x(mT - pT). \quad (19)$$

It should be noted that with a software-algorithmic solution of the problem in digital signal processing devices, it will be possible to take into account the effects of noise and eliminate them over the entire band of time intervals. In the presence of random effects, the software algorithm calculates the characteristics of random effects [8].

3 Results and discussion

The implementation of intelligent control strategies has demonstrated significant improvements in signal processing quality, particularly in reducing quantization errors and enhancing noise filtering. Simulation results of digital filters, both recursive and non-recursive, show that proper filter design allows for the minimization of transient duration and ensures accurate steady-state responses even under the influence of random disturbances.

Through analytical evaluation using Z-transform techniques, it was shown that the transfer function plays a critical role in defining system behaviour [6,7,9]. The derived relationships between input and output sequences confirmed that intelligent filtering methods can effectively suppress white noise and impulsive interference in measurement channels.

Moreover, the correlation analysis between input and output sequences revealed that the use of convolution functions allows for a deeper understanding of how random signal characteristics propagate through the system. By adjusting filter coefficients and response parameters, it becomes possible to tune the filter performance based on specific sensor dynamics and process conditions [3, 7].

Experimental testing with analogue sensors in noisy industrial environments confirmed the theoretical predictions. The application of digital filtering led to a marked improvement in measurement accuracy, while also enhancing the overall reliability of the control system by preserving the integrity of the processed data.

The study highlights the growing importance of integrating software-based algorithmic solutions within modern control system architectures. Such methods offer the adaptability required to handle diverse signal variations and enable intelligent systems to function reliably under uncertain conditions—laying the groundwork for broader implementation of AI-powered control technologies in industrial settings [2, 5, 9].

```
double arith_mean(auto assign, int buffer_size=10){
    this buffer = sizof(assign) * buffer_size;}

    //Move the buffer to the actual values ([0, 1, 2, 3] -> [1,
 2, 3, 4])
    this.buffer = this.buffer[1:];
    this.buffer.push_back(assign);

    //Calculation of the arithmetic mean
    double mean = 0;
    for(double e: this.buffer){
        mean+=e;
    }
    mean = mean/sizeof(this.buffer);

    return mean;
```

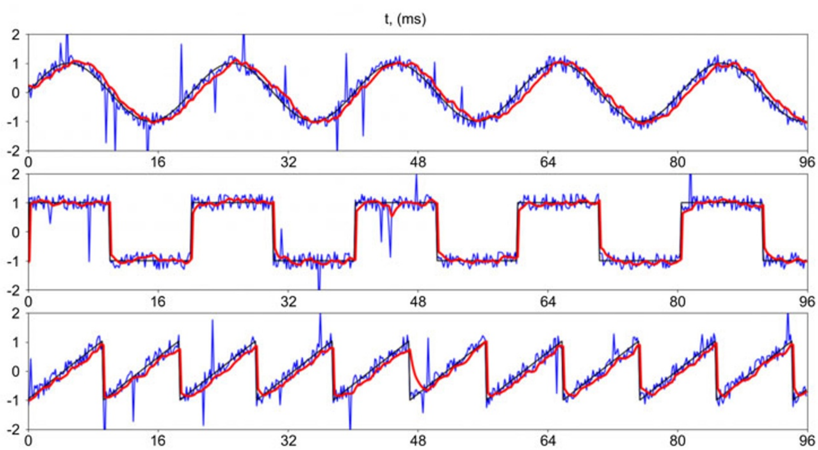

Fig. 3. Filtering of noisy input signals with various waveforms using a digital filter.

The digital filter demonstrates high performance in processing a variety of signal types. It successfully removes different forms of noise—both high-frequency and impulsive—while maintaining the structural integrity of the underlying signal. These results validate the effectiveness of the algorithmic approach for real-time signal correction in measurement and control systems [5, 8, 11].

4 Conclusion

The transition from traditional adaptive systems to intelligent control architectures marks a significant advancement in process automation. This study emphasizes that software-based algorithmic methods in digital filtering effectively reduce quantization errors and random noise, leading to improved reliability and accuracy of sensor data. Recursive and non-recursive filter models, implemented through digital signal processing, facilitate real-time analysis and correction of incoming signals [4]. By integrating intelligent technologies, such as neural networks and fuzzy control systems, future control architectures can operate more effectively in uncertain and dynamic industrial environments. This paves the way for highly responsive, adaptive, and intelligent process management in modern automation systems.

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