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"REALIZATION OF NUMERICAL FILTERS ON STM32F103 MICROCONTROLLERS"

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Realization of Numerical Filters on STM32F103 Microcontrollers

Abstract

This thesis explores the design, implementation, and optimization of numerical filters on STM32F103 microcontrollers using Keil software. Numerical filters, including low-pass and high-pass filters, are fundamental in signal processing, enabling the removal of unwanted noise and the enhancement of signal features. The STM32F103, equipped with the ARM Cortex-M3 core, offers a balance of performance and power efficiency, making it an ideal platform for embedded signal processing tasks.

In this work, we detail the development process of various numerical filters, leveraging Keil µVision for code development and debugging. The study includes the implementation of FIR and IIR filters, with a focus on optimizing their computational efficiency and memory usage. We utilize the CMSIS-DSP library to accelerate filter development and ensure adherence to industry standards. The performance of the filters is rigorously tested through simulation and real-world applications, demonstrating their effectiveness in real-time signal processing.

The results highlight the STM32F103's capability to handle complex filtering tasks with minimal resource consumption, making it suitable for a wide range of applications, from industrial automation to IoT sensor networks. The thesis concludes with recommendations for future research, emphasizing the development of user-friendly interfaces, enhanced security features, and the expansion of the filtering library to support diverse signal processing needs.

Abstract

Questa tesi esplora la progettazione, l'implementazione e l'ottimizzazione dei filtri numerici sui microcontrollori STM32F103 utilizzando il software Keil. I filtri numerici, inclusi i filtri passa-basso e passa-alto, sono fondamentali nell'elaborazione del segnale, poiché consentono la rimozione del rumore indesiderato e il miglioramento delle caratteristiche del segnale. STM32F103, dotato del core ARM Cortex-M3, offre un equilibrio tra prestazioni ed efficienza energetica, rendendolo una piattaforma ideale per attività di elaborazione del segnale embedded.

In questo lavoro, descriviamo in dettaglio il processo di sviluppo di vari filtri numerici, sfruttando Keil µVision per lo sviluppo e il debug del codice. Lo studio include l'implementazione di filtri FIR e IIR, con particolare attenzione all'ottimizzazione della loro efficienza computazionale e dell'utilizzo della memoria. Utilizziamo la libreria CMSIS-DSP per accelerare lo sviluppo dei filtri e garantire l'aderenza agli standard del settore. Le prestazioni dei filtri vengono rigorosamente testate attraverso simulazioni e applicazioni nel mondo reale, dimostrando la loro efficacia nell'elaborazione del segnale in tempo reale.

I risultati evidenziano la capacità di STM32F103 di gestire attività di filtraggio complesse con un consumo minimo di risorse, rendendolo adatto a un'ampia gamma di applicazioni, dall'automazione industriale alle reti di sensori IoT. La tesi si conclude con raccomandazioni per la ricerca futura, sottolineando lo sviluppo di interfacce user-friendly, funzionalità di sicurezza migliorate e l'espansione della libreria di filtraggio per supportare diverse esigenze di elaborazione del segnale.

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Declaration

I hereby declare that the work presented in this thesis, titled “Realization of Numerical Filters on STM32F103 Microcontrollers” was carried out by me for bachelor's thesis under the guidance and supervision of **Professor Simone BUSO**, Faculty of Engineering, Department of Electronic and information Engineering, University of Padova, Italy.

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1 INTRODUCTION

1.1. Background

Digital signal processing (DSP) plays a vital role in various applications, from audio and image processing to communications and control systems. Embedded systems, particularly those based on microcontrollers, often require efficient DSP techniques to process real-time signals. Numerical filtering is a fundamental DSP technique used to remove noise and enhance signal quality. The STM32F103 microcontroller, part of the STM32 family from STMicroelectronics, is widely used in embedded systems due to its powerful features and cost-effectiveness.

1.2. Problem Statement

Implementing efficient numerical filters on resource-constrained devices like microcontrollers presents significant challenges. These include limitations in computational power, memory constraints, and the need for real-time processing. This thesis aims to address these challenges by exploring the creation and optimization of numerical filters on the STM32F103 microcontroller.

1.3. Objectives

- To explore the theoretical foundations of numerical filters.
- To design and implement FIR and IIR filters on the STM32F103 microcontroller.
- To evaluate the performance of these filters in practical scenarios.
- To provide optimization techniques for enhancing filter performance on the STM32F103.

1.4. Scope of the Thesis

The research focuses on numerical filtering techniques, particularly FIR and IIR filters, and their implementation on the STM32F103 microcontroller. The study includes the design, coding, optimization, and performance evaluation of these filters in real-world applications.

1.5. Structure of the Thesis

The thesis is organized into seven chapters, starting with an introduction, followed by a literature review, theoretical background, detailed discussion of the STM32F103 microcontroller, the creation and implementation of numerical filters, performance evaluation, and concluding with a summary of findings and future directions.

2 LITERATURE REVIEW

2.1 Overview of Numerical Filtering

Numerical filtering in DSP refers to the process of modifying or enhancing signals through various mathematical operations. These filters can remove noise, extract important features, or transform signals for further analysis.

Filtering is a fundamental technique in DSP that enhances signal quality and extracts useful information across various applications and plays a vital role in improving the performance and accuracy of numerous systems and technologies [1].

The importance of filtering can be explained as below:

- Noise reduction is critical in many applications where the integrity of the signal must be preserved. Noise, which can be introduced by external sources or electronic components, can obscure important signal features and degrade performance.

- Signal separation involves isolating different components of a signal that are mixed together. This is important in applications where multiple signals are combined, and each component needs to be analyzed or processed separately.

- Data smoothing is used to eliminate short-term fluctuations and highlight long-term trends or cycles in data. This is crucial in applications where noisy data can obscure important patterns or trends [2].

2.2 Types of Numerical Filters: FIR and IIR

Finite Impulse Response (FIR) and Infinite Impulse Response (IIR) filters are the two primary types of filters used in digital signal processing (DSP). They differ in their characteristics, design, and applications.

FIR filters have an impulse response with limited duration. This means that the filter's response to an impulse input goes to zero within a finite time. They can also be designed to have an exact linear phase, and the response is a linear function of frequency. This ensures that all frequency components of the input signal are delayed by the same amount and the waveform is preserved. In addition, FIR filters are always stable because their poles are located at the origin in the z-plane.

FIR filters are ideal for applications where phase linearity is critical, such as data communications and image processing. They ensure stability because they do not rely on feedback and can be designed to have a precise linear phase. However, typically require more multiplications and additions compared to IIR filters for the same performance and often need more memory to store filter coefficients and intermediate calculations [3].

On the other hand, IIR (Infinite Impulse Response) Filters have an impulse response that theoretically lasts forever, allowing them to achieve the desired response with fewer coefficients using feedback from the output to the input. The feedback mechanism can lead to instability if not carefully designed. IIR filters have a non-linear phase response that can distort signals.

The advantage of IIR filters is the efficiency that allows us to obtain a certain frequency response with fewer coefficients than FIR filters and requires less memory for the filter coefficients. However, they can become unstable if not designed correctly and distort phase-sensitive signals [4].

In conclusion, FIR and IIR filters serve different needs and have unique strengths and weaknesses, making each more suitable for specific applications.

FIR filters are simpler to implement because they do not require feedback and are inherently stable. IIR filters, while more efficient in terms of the number of computations, require careful design to ensure stability.

FIR filters can achieve exact linear phase, making them suitable for applications where phase characteristics are critical. IIR filters usually have non-linear phase characteristics, which can distort phase-sensitive signals.

The choice between FIR and IIR filters depends on the specific requirements of the application. FIR filters are preferred for applications requiring precise phase control and guaranteed stability, such as in audio and communication systems. IIR filters are advantageous in scenarios where computational efficiency is paramount and phase linearity is less critical, such as in general-purpose signal processing tasks. The design of these filters involves trade-offs between computational cost, stability, phase response, and memory usage, necessitating a careful assessment of the application's demands [5].

2.3 Microcontrollers in DSP Applications

Digital signal processing (DSP) is a critical technology in modern electronics that enables the manipulation and analysis of signals in various applications such as audio processing, communications, and control systems. Microcontrollers (MCU) have become a popular choice for performing DSP tasks in embedded systems due to their affordability, versatility, and energy efficiency.

Microcontrollers play a vital role in embedded systems and provide an efficient and cost-effective platform for performing DSP tasks. Their integration, low power consumption and versatility make them suitable for a wide range of applications, from consumer electronics to industrial automation. Despite some limitations in performance and memory, the advantages of using microcontrollers in DSP tasks, such as ease of development and system integration, outweigh the challenges. As technology advances, the capabilities of microcontrollers continue to expand, reinforcing their importance in embedded DSP applications [6].

2.4 STM32F103 Microcontroller: Architecture and Features

The STM32F103 microcontroller is part of the STM32F1 series from STMicroelectronics, based on the ARM Cortex-M3 core. This microcontroller is designed for high performance, low power consumption, and advanced peripherals, making it suitable for a variety of applications.

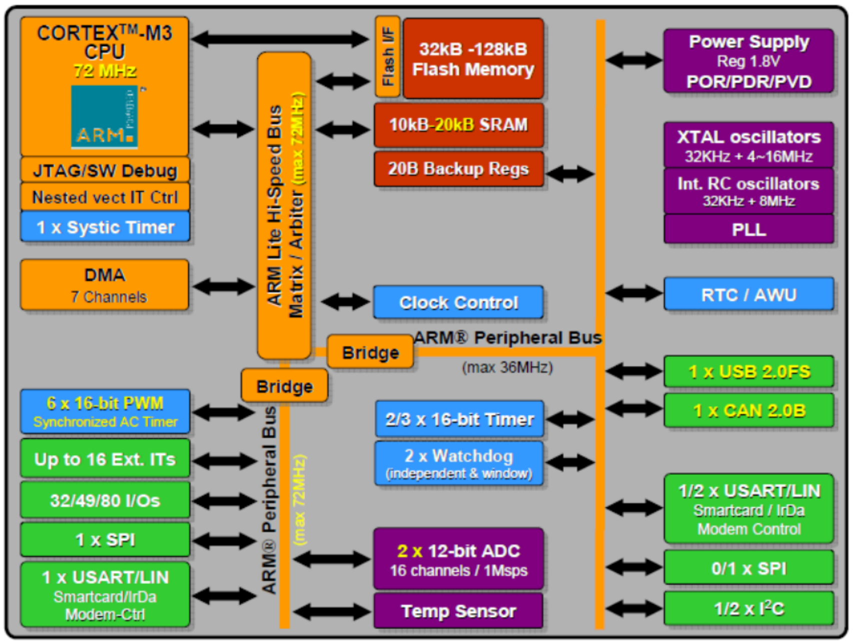


Fig. 2.1 STM32F103 Performance line Block Diagram

Key Components are:

- **Core and Performance**:
  - ARM Cortex-M3 32-bit processor core includes a hardware MAC unit; which is essential for efficient implementation of DSP algorithms like FIR and IIR filters.
  - Up to 72 MHz CPU frequency.
  - Nested Vectored Interrupt Controller (NVIC) for efficient interrupt handling.
  - Single-cycle multiply and hardware divide.

- **Memory**:
  - Up to 128 KB of on-chip Flash memory.
  - Up to 20 KB of SRAM.

- **Clock and Reset**:
  - Internal RC oscillators (8 MHz and 40 kHz).
  - External clock sources (4-16 MHz crystals/ceramic resonators).
  - Phase-Locked Loop (PLL) for clock generation.
  - Low-power modes: Sleep, Stop, and Standby.

- **Digital I/O**:

- Up to 80 I/O ports, configurable as general-purpose I/O or peripheral functions.
- Programmable output speed (2 MHz, 10 MHz, 50 MHz).

- **Analog Peripherals**:
  - 12-bit ADCs: Up to 16 channels with up to 1 Msps conversion rate.
  - Digital-to-Analog Converters (DAC): Not available in all models.
  - Comparators: Not available in all models.

- **Communication Interfaces**:
  - USARTs: Up to 5, supporting synchronous/asynchronous communication.
  - SPI: Up to 3, supporting full-duplex synchronous communication.
  - I2C: Up to 2, supporting multi-master/slave modes.
  - CAN: Up to 2, for automotive and industrial communication.
  - USB: Full-speed USB 2.0 interface.

- **Timers**:
  - Advanced-control timers: 4x 16-bit timers with PWM and input capture/compare.
  - General-purpose timers: 4x 16-bit timers.
  - Basic timers: 2x for timing and DAC triggering.

- **DMA Controller:**
  - 7-channel DMA controller for efficient data transfer without CPU intervention.

The STM32F103 microcontroller offers a robust set of features and capabilities that make it well-suited for embedded DSP applications. Its ARM Cortex-M3 core, combined with flexible peripherals and low-power modes, enables efficient and effective DSP operations in a variety of applications. While it may not match the performance of dedicated DSP processors, its versatility and integration make it an excellent choice for many embedded system designs [7].

2.5 Previous Work on Numerical Filtering in Embedded Systems

Digital filters are essential components in digital signal processing (DSP) that are used to filter out unwanted components from a signal. Various microcontrollers (MCU) have been used to implement these filters, and several studies have investigated their efficiency, techniques, and limitations.

One notable study involved the STM32F407 microcontroller, a member of the STM32 family by STMicroelectronics. Researchers designed and implemented FIR and IIR filters using ARM's CMSIS DSP libraries. The study highlighted the advantages of using hardware-optimized libraries to enhance the performance of numerical filters. FIR filters were implemented using windowing methods, while IIR filters employed bilinear transformation techniques. The findings revealed that, despite the computational overhead associated with FIR filters due to their larger number of coefficients, the STM32F407 managed these tasks effectively, although IIR filters were preferred in scenarios requiring lower computational loads and memory usage [8].

In the realm of Arduino microcontrollers, a study explored real-time DSP capabilities using the Arduino Uno. The implementation focused on basic FIR and IIR filters, programmed using simple C code without hardware acceleration. The direct form was used for FIR filters, while biquad structures were chosen for IIR filters. The results demonstrated that while Arduino Uno, with its 8-bit architecture, could handle basic DSP tasks, its limited processing power and memory posed significant constraints. These limitations were particularly evident when dealing with higher-order filters and high sampling rates, suggesting that Arduino is better suited for low-complexity, cost-sensitive applications.

The ESP32 microcontroller, known for its dual-core 32-bit architecture, was investigated for its DSP capabilities in IoT applications. Utilizing the ESP-DSP library, researchers implemented both FIR and IIR filters, leveraging the microcontroller's hardware capabilities to optimize performance. The study found that the ESP32 could handle real-time DSP tasks efficiently, making it suitable for IoT applications requiring significant processing power. However, the power consumption remained a critical consideration, especially for battery-powered devices [9].

A study on the PIC32MX series microcontrollers focused on implementing digital filters using the Microchip Libraries for Applications (MLA). Both FIR and IIR filters were implemented, demonstrating the microcontroller's ability to handle moderate DSP tasks effectively. The study emphasized the importance of using assembly code for critical sections to optimize performance, addressing the limitations in processing speed observed with high-level programming languages.

For biomedical applications, the PIC24 microcontroller was used to design IIR filters for processing biomedical signals such as ECG data. The filters were implemented using a direct form II structure to minimize memory usage and enhance stability. This implementation underscored the PIC24's suitability for medical applications, where low power consumption and efficient processing are paramount. However, challenges related to processing delays and power management highlighted the need for further optimization, particularly for portable, battery-operated medical devices [10].

These studies collectively illustrate the diverse capabilities and constraints of different microcontrollers in executing numerical filters. The STM32 family stands out for its balance of performance and flexibility, suitable for a wide range of applications including audio processing and moderate DSP tasks. Arduino microcontrollers, while limited in processing power and memory, offer a low-cost solution for simple DSP applications. The ESP32 provides robust performance for IoT applications, leveraging its dual-core processing capabilities, though power consumption remains a concern. Meanwhile, PIC microcontrollers are well-suited for specific tasks like biomedical signal processing, provided that performance optimization techniques are employed to overcome inherent limitations.

3 THEORY OF NUMERICAL FILTERS

3.1 Fundamentals of Digital Signal Processing

Digital Signal Processing (DSP) is an essential field in modern electronics that deals with the manipulation of digital signals to extract or alter information. Several key concepts form the foundation of DSP, including sampling, quantization, and the Z-transform.

Sampling is the process of converting a continuous-time signal into a discrete time signal by taking measurements at regular intervals. This process is governed by the Nyquist-Shannon sampling theorem, which states that to accurately reconstruct the original signal, the sampling rate must be at least twice the highest frequency present in the signal. If the sampling rate is too low, aliasing can occur, where higher frequency components are incorrectly mapped to lower frequencies, leading to distortion.

Quantization is the process of mapping a continuous range of signal values to a finite range of discrete levels. After sampling, each analog sample is assigned a digital value from a limited set of levels, which introduces a quantization error. This error is the difference between the actual analog value and the quantized digital value. Quantization can be uniform or non-uniform. Uniform quantization assigns equal step sizes between levels, while non-uniform quantization uses varying step sizes, often to give more precision to more critical signal ranges. The resolution of the quantization process is determined by the number of bits used for each sample; higher bit-depth results in smaller quantization errors and better signal fidelity [11].

The Z-transform is a mathematical tool used in DSP to analyze and design digital systems. It is a generalization of the discrete-time Fourier transform (DTFT) and is particularly useful for dealing with linear, time-invariant (LTI) systems. The Z-transform converts a discrete-time signal, which is a sequence of numbers, into a complex frequency domain representation. This transformation simplifies the analysis of complex signals and systems, making it easier to understand their behavior and design filters. The Z-transform of a discrete signal x[n] is defined as X(z) = Σ x[n]z^(-n), where z is a complex variable. Key concepts related to the Z-transform include the region of convergence (ROC), which defines where the Z-transform converges, and the inverse Z-transform, which is used to convert the signal back to the time domain.

Together, these concepts enable the effective processing of digital signals. Sampling allows continuous signals to be represented in a digital form, quantization ensures these digital representations are manageable within digital systems, and the Z-transform provides a powerful method for analyzing and designing signal processing algorithms. Understanding these fundamental principles is crucial for anyone working in the field of DSP, as they form the basis for more advanced techniques and applications in communications, audio processing, control systems, and beyond.

3.2 Peripherals and DSP Capabilities

The STM32F103 microcontroller, part of STMicroelectronics STM32F1 series, is built around the ARM Cortex-M3 core, which is known for its balanced performance and power efficiency. This microcontroller integrates a variety of peripherals that enhance its capabilities for digital signal processing (DSP), making it suitable for implementing numerical filters.

Timers in the STM32F103 are versatile and numerous, providing essential functions for timing control, signal generation, and event management. Advanced-control timers offer features like pulse-width modulation (PWM) and input capture/compare, which are crucial for applications requiring precise timing and signal generation. General-purpose timers can be used for simpler timing tasks, and basic timers serve well for straightforward timekeeping. These timers facilitate the implementation of numerical filters by providing accurate sampling intervals, which are critical for maintaining the integrity of the sampled signals during filtering operations.

The STM32F103 also features Analog-to-Digital Converters (ADCs), which play a pivotal role in DSP by converting analog signals into digital form. With up to 16 channels and a 12-bit resolution, the ADCs can achieve sampling rates up to 1 million samples per second (Msps). This high sampling rate and resolution ensure that the analog signals are captured with sufficient detail and speed, allowing for effective digital filtering and signal processing. The ADCs can be configured to operate in different modes, including single, continuous, scan, and discontinuous modes, providing flexibility in how signals are sampled and processed.

Although not available in all models, Digital-to-Analog Converters (DACs) can be found in some STM32F103 variants. DACs convert digital signals back into analog form, which is essential for applications where the processed digital signals need to be output as analog signals. This capability is particularly useful in audio processing applications, where the filtered digital audio signals are converted back into analog form for playback [12].

Dedicated DSP instructions supported by the ARM Cortex-M3 core further enhance the STM32F103's capabilities for signal processing. The Cortex-M3 includes hardware support for single-cycle multiply and multiply-accumulate (MAC) operations, which are fundamental to many DSP algorithms, such as finite impulse response (FIR) and infinite impulse response (IIR) filters. The MAC operation, in particular, is critical for efficient filter implementation, as it allows for rapid computation of the dot products involved in FIR and IIR filtering. Additionally, the core supports saturation arithmetic, which helps prevent overflow in fixed-point calculations, a common issue in DSP applications.

These DSP features and peripherals collectively facilitate the implementation of numerical filters on the STM32F103. The timers ensure precise control over sampling intervals, which is vital for maintaining the consistency and accuracy of the filtered signals. The high-resolution and high-speed ADCs allow for detailed and rapid digitization of analog signals, providing the raw data necessary for digital filtering. DACs enable the conversion of processed digital signals back into analog form, ensuring that the final output can be utilized in analog domains. The dedicated DSP instructions, particularly the MAC operations, significantly speed up the computation of filter algorithms, making real-time processing feasible even for complex filters.

In summary, the combination of versatile timers, high-performance ADCs, available DACs, and dedicated DSP instructions in the STM32F103 microcontroller provides a robust platform for implementing numerical filters. These features enable precise sampling, efficient signal conversion, and rapid computation, all of which are essential for effective digital signal processing in a wide range of applications [13].

3.3 Filter Design Considerations for Microcontrollers

Designing and implementing filters on microcontrollers involves a set of unique considerations due to the constraints and characteristics of these systems. It's essential to account for the limited computational power, memory constraints, and real-time processing requirements typical of microcontrollers. Understanding the specific needs of the application, such as the type of signals being processed and the desired filter characteristics, is crucial [14].

When implementing filters, the choice between hardware and software implementations must be made based on factors like precision requirements, processing speed, and resource availability. Software implementations might rely on fixed-point arithmetic to optimize performance, considering that many microcontrollers lack floating-point units. Efficient coding practices, such as minimizing loop overhead and optimizing mathematical operations, are vital to ensure that the filter operates within the real-time constraints.

Another key consideration is the power consumption of the filter algorithm, as microcontrollers are often used in battery-powered devices. Algorithms need to be energy-efficient to prolong battery life. Additionally, the implementation should be robust against numerical stability issues, especially in recursive filters like IIR filters.

Finally, testing and validation of the filter implementation are crucial to ensure it meets the performance criteria under all operating conditions. This includes verifying the filter's frequency response, phase response, and overall stability within the microcontroller's operational limits. Properly addressing these considerations ensures that the filter performs effectively and reliably in the intended application [15].

3.4 Development Environment and Tools

When programming the STM32F103 microcontroller, developers have access to a range of software tools and development environments that cater to different needs and preferences. A primary tool is STM32CubeIDE, an integrated development environment from STMicroelectronics. It combines STM32CubeMX's graphical configuration and code generation with the advanced features of the Eclipse-based IDE, providing a comprehensive suite for project management, code editing, and debugging. Its integration with STM32CubeMX simplifies peripheral configuration and initialization code generation, enhancing development efficiency.

Another key resource is the CMSIS-DSP library, which offers optimized signal processing algorithms specifically for ARM Cortex-M processors. This library includes functions for

FFTs, filters, matrix operations, and more, supporting both fixed-point and floating-point implementations. Its seamless integration with ARM's CMSIS standard ensures compatibility and ease of use, making it ideal for complex signal processing tasks [16].

For developers seeking an alternative IDE, Keil MDK and IAR Embedded Workbench are popular choices. Keil MDK features the µVision IDE, which excels in debugging with advanced features like trace, profiling, and performance analysis, along with RTOS support and middleware components. IAR Embedded Workbench is known for its high-performance compiler that optimizes code size and execution speed, coupled with robust debugging tools and RTOS-aware capabilities.

System Workbench for STM32 (SW4STM32) and ARM Mbed Studio offer additional options. SW4STM32 is an Eclipse-based IDE provided by AC6, valued for its free and open-source nature, making it accessible for a wide range of users. It also integrates with STM32CubeMX for graphical configuration. ARM Mbed Studio focuses on IoT applications, integrating closely with Mbed OS and providing cloud connectivity features, which is beneficial for IoT and embedded applications.

The GNU Arm Embedded Toolchain (GCC) is another essential tool, being open-source and supported by a large community. It offers flexibility, allowing use with various IDEs and build systems like Eclipse, Makefiles, and CMake, and includes a robust optimizing compiler.

These tools collectively provide a versatile and powerful ecosystem for developing on the STM32F103 microcontroller. The choice of tool often depends on specific project requirements, such as the need for advanced debugging, optimization capabilities, ease of peripheral configuration, or support for IoT functionalities. Each tool and environment brings its own strengths, enabling developers to efficiently tackle diverse embedded system challenges [17].

3.5 Challenges in Implementing Numerical Filters on STM32F103

Implementing numerical filters on the STM32F103 microcontroller presents several specific challenges due to its inherent constraints and characteristics. One of the primary challenges is the limited computational power of the STM32F103, which typically features an ARM Cortex-M3 core. While this core is efficient for many embedded tasks, it lacks the processing muscle of more advanced processors, making it necessary to carefully optimize filter algorithms to run efficiently within the available CPU cycles.

Memory constraints also pose a significant challenge. The STM32F103 typically has limited RAM and flash memory, which can restrict the size and complexity of filter implementations. Efficient memory management becomes crucial, necessitating the use of fixed-point arithmetic instead of floating-point to save memory and processing power. However, using fixed-point arithmetic introduces its own complexities, such as scaling issues and the need for careful handling of numerical precision to avoid overflow and underflow errors [18].

Real-time processing requirements further complicate filter implementation. Many applications using the STM32F103 need to process data in real-time, necessitating filters that can execute within strict timing constraints. This means the filter algorithms must be highly optimized to ensure they can handle the incoming data rate without introducing unacceptable latency or missing deadlines. Achieving this level of performance often requires a deep understanding of both the hardware and the specific requirements of the application.

Power consumption is another critical consideration, especially in battery-powered applications. Efficient filter implementation must balance the need for computational accuracy and performance with the need to minimize energy consumption. This often involves making trade-offs between algorithm complexity and power usage and implementing techniques such as power-saving modes and efficient use of the processor's sleep states [19].

Moreover, implementing recursive filters, like Infinite Impulse Response (IIR) filters, requires special attention to numerical stability. The limited precision of fixed-point arithmetic can lead to stability issues over time, necessitating careful design and testing to ensure the filter remains stable under all operating conditions. This often involves implementing safeguards and error-checking mechanisms to detect and mitigate potential stability problems.

Lastly, the development and debugging tools available for the STM32F103 can also influence the implementation of numerical filters. While tools like STM32CubeIDE and the CMSIS-DSP library provide significant support, developers still need to ensure that their development environment is properly configured to handle the specific requirements of numerical filter implementation. This includes setting up efficient workflows for testing and validating the filter performance under real-world conditions.

In summary, implementing numerical filters on the STM32F103 involves navigating challenges related to computational power, memory constraints, real-time processing, power consumption, numerical stability, and the effective use of development tools. Addressing these challenges requires a careful and well-informed approach to algorithm design, optimization, and testing to ensure that the filters perform reliably and efficiently within the microcontroller's limitations [20].

3.6 Overview of the Keil µVision 5 development environment

Keil µVision 5 IDE offers a comprehensive environment tailored for embedded systems development, particularly for ARM microcontrollers. When setting up a project, users benefit from an intuitive interface that simplifies the creation and management of projects. It supports various templates and device configurations, making it easier to get started with different microcontrollers, including those from the STM32 family [21].

In the realm of code editing, Keil µVision 5 provides a powerful editor with features like syntax highlighting, code completion, and context-sensitive help. These features enhance productivity by reducing common coding errors and speeding up the development process. The editor also integrates well with the debugging tools, allowing for seamless transition between writing and testing code.

Compiling in Keil µVision 5 is streamlined through its robust build system, which supports various optimization levels and linker settings. The IDE generates efficient machine code optimized for performance and memory usage, essential for embedded applications where resources are often limited.

Debugging is one of the standout features of Keil µVision 5. The IDE includes an advanced debugging interface with capabilities such as breakpoints, watch windows, memory views, and real-time variable tracking. These tools are crucial for diagnosing and fixing issues in embedded systems. The integrated simulator allows for testing and debugging code even without the actual hardware, which is particularly useful during the early stages of development.

Interfacing with STM32CubeMX, Keil µVision 5 excels in peripheral configuration and project management. STM32CubeMX, a graphical configuration tool, helps in setting up peripherals, pin configurations, and middleware for STM32 microcontrollers. The integration with Keil µVision 5 ensures that once the configuration is done in STM32CubeMX, the settings and initialization code can be seamlessly imported into the IDE. This saves time and reduces the likelihood of errors in manual configuration, ensuring that the peripheral setup is consistent with the code being developed.

Overall, Keil µVision 5 IDE offers a comprehensive set of tools for embedded systems development, from project setup and code editing to compiling, debugging, and advanced peripheral configuration through STM32CubeMX. This integration and feature set make it a powerful choice for developers working with ARM microcontrollers, particularly those in the STM32 family [22].

4 IMPLEMENTATION OF NUMERICAL FILTERS

4.1 Test setup and hardware connections

To implement numerical filters on the STM32F103 microcontroller using Keil µVision 5, specific hardware components to set up, test, and validate the filter algorithms are needed. Such as:

- **STM32F103 Microcontroller**
- **Development Board**: An STM32F103-based development board, such as the STM32F103C8T6 "Blue Pill" or Nucleo-64 board, which includes essential components like power regulation and basic I/O pins.
- **ST-Link/V2 Debugger and Programmer**: For flashing the firmware and debugging the microcontroller.
- **USB Cable**: To connect the development board to the computer for power, programming, and serial communication.
- **Push Buttons and Switches**: For user input.
- **LEDs and Resistors**: For basic output indicators.
- **SPI Modules**: For communication with peripherals like sensors or displays.
- **LCD or OLED Display**: For visual output and debugging information.

These components provide a solid foundation for developing and testing applications with the STM32F103 microcontroller.

4.2 Configuration of STM32F103 peripherals (ADC, Timers, UART)

Configuring ADC, timers, and UART using STM32CubeMX and integrating these configurations into a Keil project involves several steps that blend graphical configuration with code integration.

To start, we launched STM32CubeMX and we created a new project. STM32 microcontroller by selecting its model number has been chosen. Once the project is initialized, we are with the pinout view, where we can configure the necessary peripherals.

For configuring the ADC, we should click on the pin associated with the ADC input and set it to analog mode. Then in "Configuration" tab, we should select the ADC peripheral, and set its parameters such as resolution, data alignment, and conversion mode. We can also add multiple channels and set the sampling time for each channel.

To configure timers, we can enable the desired timer by clicking on its associated pin. In the "Configuration" tab, we set its mode (e.g., PWM, input capture, output compare). Then, we adjust the prescaler and counter period to achieve the desired frequency, and configure the timer channels if needed [23].

For UART configuration, we enable the UART peripheral (e.g., USART2) from the pinout view. In the "Configuration" tab, we set the UART parameters such as baud rate, word length, stop bits, parity, and mode (TX, RX, or both). This ensures the UART is correctly set up for communication.

After configuring all peripherals, we generated the initialization code by clicking "Project" in the top menu and we chose the MDK-ARM toolchain for Keil and generate the code. STM32CubeMX will create a project directory with all the necessary files, including main.c, stm32fxxx_hal_msp.c, stm32fxxx_it.c, and stm32fxxx_hal_conf.h.

In Keil µVision we can load the generated project by opening the .uvprojx file in the project directory. STM32CubeMX generates several files, including:

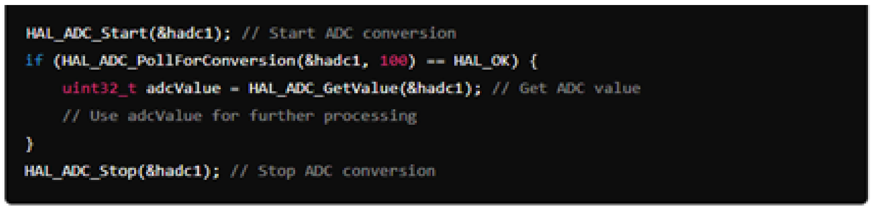
- main.c – Main program.
- stm32fxxx_hal_msp.c – MSP (MCU Support Package) initialization code.
- stm32fxxx_it.c – Interrupt service routines.
- stm32fxxx_hal_conf.h – HAL configuration file.

In main.c, user code sections marked with USER CODE BEGIN and USER CODE END can be found. These sections are preserved when regenerating code from STM32CubeMX and we safely add our application-specific code here.

To use the configured peripherals in our application, include the necessary HAL library functions.

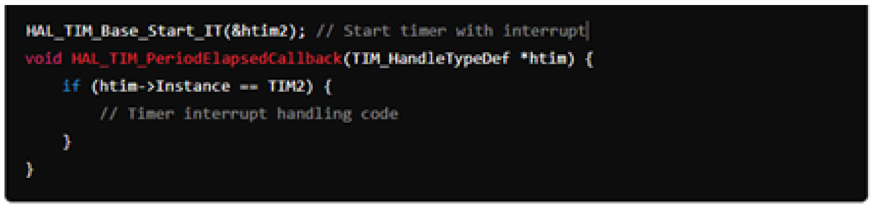
Here's a simple example of how to use the configured ADC, timer, and UART:

ADC Example:



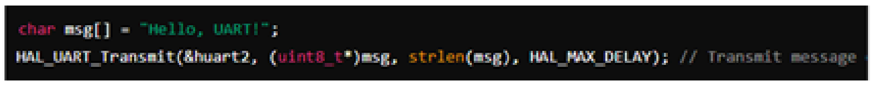
```
HAL_ADC_Start(&hadc1); // Start ADC conversion
if (HAL_ADC_PollForConversion(&hadc1, 100) == HAL_OK) {
    uint32_t adcValue = HAL_ADC_GetValue(&hadc1); // Get ADC value
    // Use adcValue for further processing
}
HAL_ADC_Stop(&hadc1); // Stop ADC conversion
```

Timer Example:



```
HAL_TIM_Base_Start_IT(&htim2); // Start timer with interrupt
void HAL_TIM_PeriodElapsedCallback(TIM_HandleTypeDef *htim) {
    if (htim->Instance == TIM2) {
        // Timer interrupt handling code
    }
}
```

UART Example:



```
char msg[] = "Hello, UART!";
HAL_UART_Transmit(&huart2, (uint8_t*)msg, strlen(msg), HAL_MAX_DELAY); // Transmit message
```

We can compile the project in Keil by clicking the "Build" button. Connect STM32 microcontroller to our development PC using a debugger (e.g., ST-LINK) and load the compiled program onto the microcontroller. Start a debug session to test and troubleshoot our application.

By following these steps, we can effectively configure ADC, timers, and UART using STM32CubeMX and integrate these configurations into a Keil project. This process leverages the strengths of both tools, making peripheral configuration straightforward and ensuring a smooth transition to application development.

4.3 Implementation of numerical filters in C

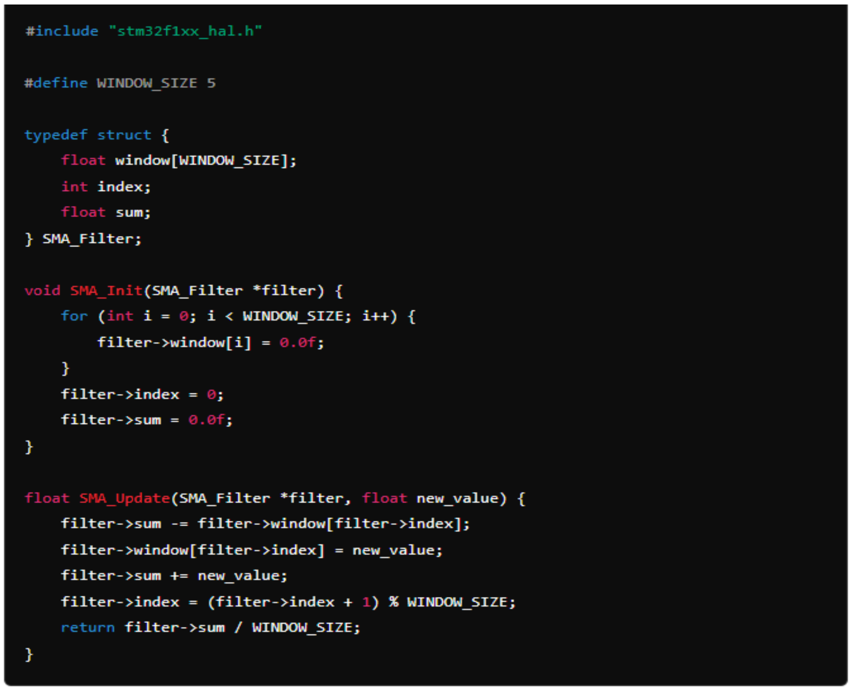
Coding numerical filters such as the Simple Moving Average (SMA), Exponential Moving Average (EMA), and Low-Pass Filter (LPF) for an STM32F103 microcontroller involves understanding the mathematical basis of each filter and translating this into efficient C code that runs on the microcontroller. Here below we explained the process, including initialization, updating filter values, and integration within an STM32F103 project using Keil µVision 5 [24].

4.3.1 Simple Moving Average (SMA)

The Simple Moving Average (SMA) filter smooths data by averaging a fixed number of the most recent data points. This is achieved by maintaining a circular buffer of the latest N values and updating the sum and average as new values are added.

First, we define a structure to hold the filter state, including the window of values, the current index, and the sum of the values. The initialization function sets the initial state, and the update function adds a new value, updates the sum, and computes the new average.

Here's the implementation of the SMA filter in C:

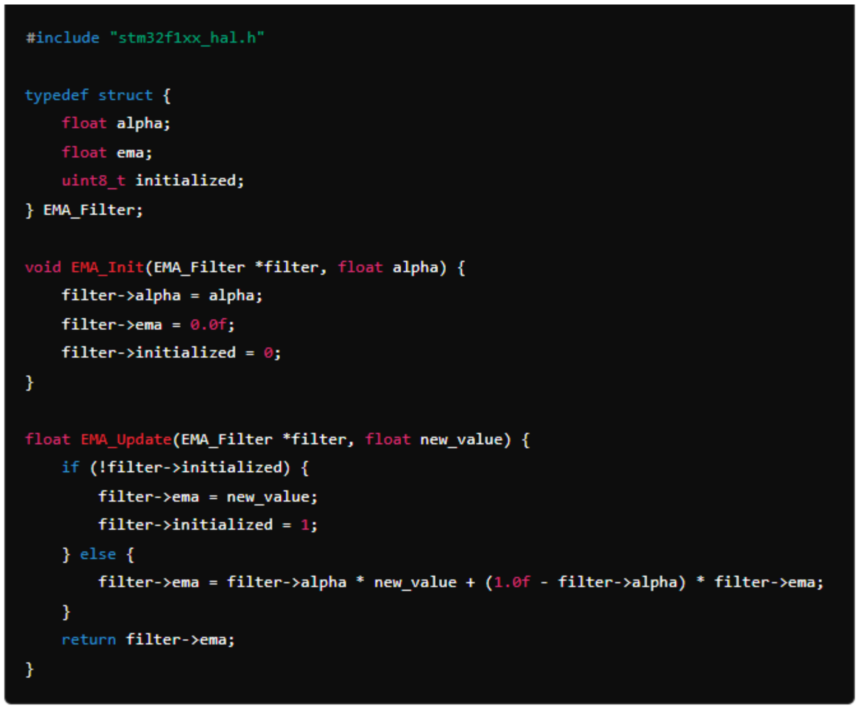


```c
#include "stm32f1xx_hal.h"

#define WINDOW_SIZE 5

typedef struct {
    float window[WINDOW_SIZE];
    int index;
    float sum;
} SMA_Filter;

void SMA_Init(SMA_Filter *filter) {
    for (int i = 0; i < WINDOW_SIZE; i++) {
        filter->window[i] = 0.0f;
    }
    filter->index = 0;
    filter->sum = 0.0f;
}

float SMA_Update(SMA_Filter *filter, float new_value) {
    filter->sum -= filter->window[filter->index];
    filter->window[filter->index] = new_value;
    filter->sum += new_value;
    filter->index = (filter->index + 1) % WINDOW_SIZE;
    return filter->sum / WINDOW_SIZE;
}
```

4.3.2 Exponential Moving Average (EMA)

The Exponential Moving Average (EMA) filter gives more weight to recent data points, providing a more responsive filter than the SMA. The EMA is defined recursively, with each new value contributing to the average in proportion to a smoothing factor alpha.

The EMA filter structure includes the alpha value and the current EMA value. The initialization function sets the initial EMA value to zero or the first input value, and the update function calculates the new EMA value based on the latest input.



```c
#include "stm32f1xx_hal.h"

typedef struct {
    float alpha;
    float ema;
    uint8_t initialized;
} EMA_Filter;

void EMA_Init(EMA_Filter *filter, float alpha) {
    filter->alpha = alpha;
    filter->ema = 0.0f;
    filter->initialized = 0;
}

float EMA_Update(EMA_Filter *filter, float new_value) {
    if (!filter->initialized) {
        filter->ema = new_value;
        filter->initialized = 1;
    } else {
        filter->ema = filter->alpha * new_value + (1.0f - filter->alpha) * filter->ema;
    }
    return filter->ema;
}
```

Here's the implementation of the EMA filter in C:

4.3.3 Low-Pass Filter (LPF)

Low-Pass Filter (LPF) on an STM32F103 microcontroller is a digital filter used to allow low-frequency signals to pass through while blocking or attenuating high-frequency signals. This type of filter is useful for smoothing out a signal, removing high-frequency noise, and extracting the low-frequency components of a signal, which are often of interest in many applications.

To implement an LPF on the STM32F103, we can configure the microcontroller's ADC (Analog-to-Digital Converter) to read an input signal and the DAC (Digital-to-Analog Converter) to output the filtered signal. The filter itself can be implemented using a difference equation. For a simple first-order low-pass filter, this equation is:

$$Y[n]=\alpha \cdot x[n]+(1-\alpha)\cdot y[n-1] \qquad (1)$$

α is a filter coefficient that determines the cutoff frequency of the filter.
The cutoff frequency determines which frequencies are allowed to pass through the filter. By adjusting the coefficient α, we can change the cutoff frequency. Lower values of α will pass lower frequencies and block higher frequencies more effectively.

The process involves continuously reading the input signal through the ADC, applying the low-pass filter algorithm, and then outputting the filtered signal via the DAC. This allows the STM32F103 to process signals in real-time, filtering out high-frequency noise and allowing the low-frequency components to be analyzed or used further in the application. This is useful in applications such as audio processing, signal conditioning, and noise reduction, where it is important to focus on the low-frequency components of a signal [25].

4.3.4 High-Pass Filter (HPF)

High-Pass Filter (HPF) on an STM32F103 microcontroller is a digital filter used to allow high-frequency signals to pass through while blocking or attenuating low-frequency signals. This type of filter is particularly useful for removing unwanted low-frequency components from a signal, such as DC offset or slow-changing trends, leaving behind the higher-frequency components that are often of more interest.

To implement an HPF on the STM32F103, we can configure the microcontroller's ADC (Analog-to-Digital Converter) to read an input signal, and the DAC (Digital-to-Analog Converter) to output the filtered signal. The filter itself can be implemented using a difference equation. For a simple first-order high-pass filter, this equation is:

$$y[n]= \beta\cdot(y[n-1]+x[n]-x[n-1]) \qquad (2)$$

β is a filter coefficient that determines the cutoff frequency of the filter.
The cutoff frequency determines which frequencies are allowed to pass through the filter. By adjusting the coefficient β, we can change the cutoff frequency. Higher values of β will pass higher frequencies and block lower frequencies more effectively.

The process involves continuously reading the input signal through the ADC, applying the high-pass filter algorithm, and then outputting the filtered signal via the DAC. This allows the STM32F103 to process signals in real-time, filtering out low-frequency components and allowing higher-frequency components to be analyzed or used further in the application. This is useful in applications such as audio processing, signal conditioning, and noise reduction, where it is important to focus on the high-frequency components of a signal [26].

4.4 Integration with Keil µVision 5

To integrate these filters into an STM32F103, begin by setting up the required peripherals using STM32CubeMX. This involves configuring the ADC for reading sensor values and the UART for debugging or data communication that is explained already in section 4.2.

After generating the initialization code in STM32CubeMX, we open the project in Keil µVision 5. We create new source and header files (e.g., filters.c and filters.h) to include the filter implementations. Including these files in our Keil project to be sure that are referenced in the main application file (main.c).

In main.c, we initialize the filters and periodically read sensor values using the ADC and apply these values to the filters and optionally send the filtered results over UART for verification. **The example of integrating the filters in main.c is available in Appendix.**

4.5 Procedures for testing the filter implementation

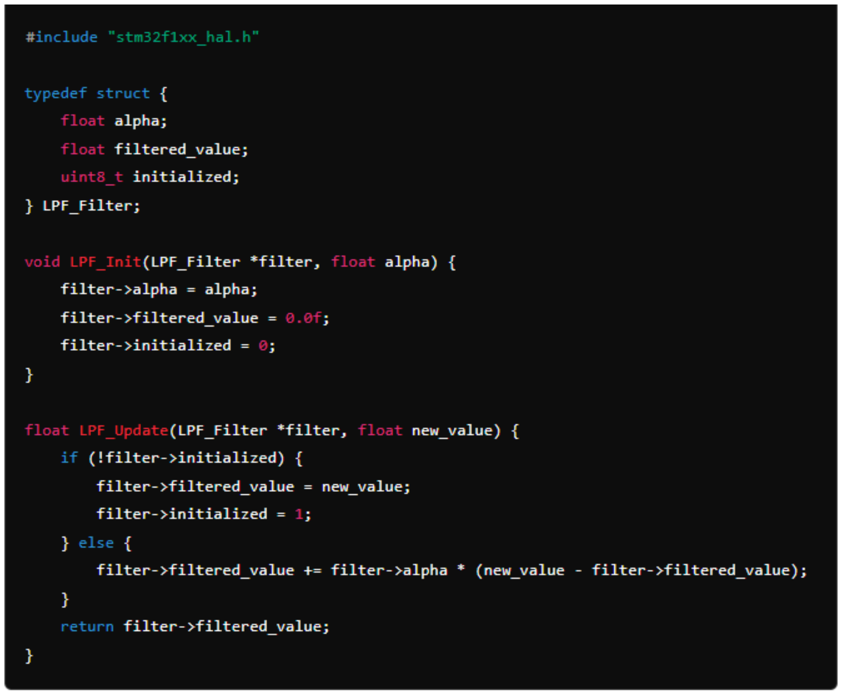
First we set up the hardware by connecting the STM32F103 microcontroller to a power source and connecting the ADC input to a signal source such as a potentiometer or a signal generator. We used a UART-to-USB adapter to connect the UART output to a PC for monitoring the filter outputs.

In the software, we create a series of test inputs, such as constant signals, step signals, and sinusoidal signals, to evaluate the performance of the filters.

4.6 Test results of LPF and HPF on real-time

In the context of Keil's implementation of a Low-Pass Filter (LPF), the alpha (α) parameter dictates the balance between smoothing the signal (filtering out high-frequency noise) and allowing the signal to closely follow rapid changes (minimizing filtering).

Here's the implementation of the LPF in C:



```c
#include "stm32f1xx_hal.h"

typedef struct {
    float alpha;
    float filtered_value;
    uint8_t initialized;
} LPF_Filter;

void LPF_Init(LPF_Filter *filter, float alpha) {
    filter->alpha = alpha;
    filter->filtered_value = 0.0f;
    filter->initialized = 0;
}

float LPF_Update(LPF_Filter *filter, float new_value) {
    if (!filter->initialized) {
        filter->filtered_value = new_value;
        filter->initialized = 1;
    } else {
        filter->filtered_value += filter->alpha * (new_value - filter->filtered_value);
    }
    return filter->filtered_value;
}
```

α = 0 (Maximum Filtering): This means the output is heavily filtered, relying almost entirely on the previous output with applying a constant voltage (e.g., 500 mV) to the ADC. The filter outputs should stabilize at or near this voltage, demonstrating their ability to handle steady-state signals.

α = 1 (Minimum Filtering): This means the output follows the input exactly, with no filtering applied.

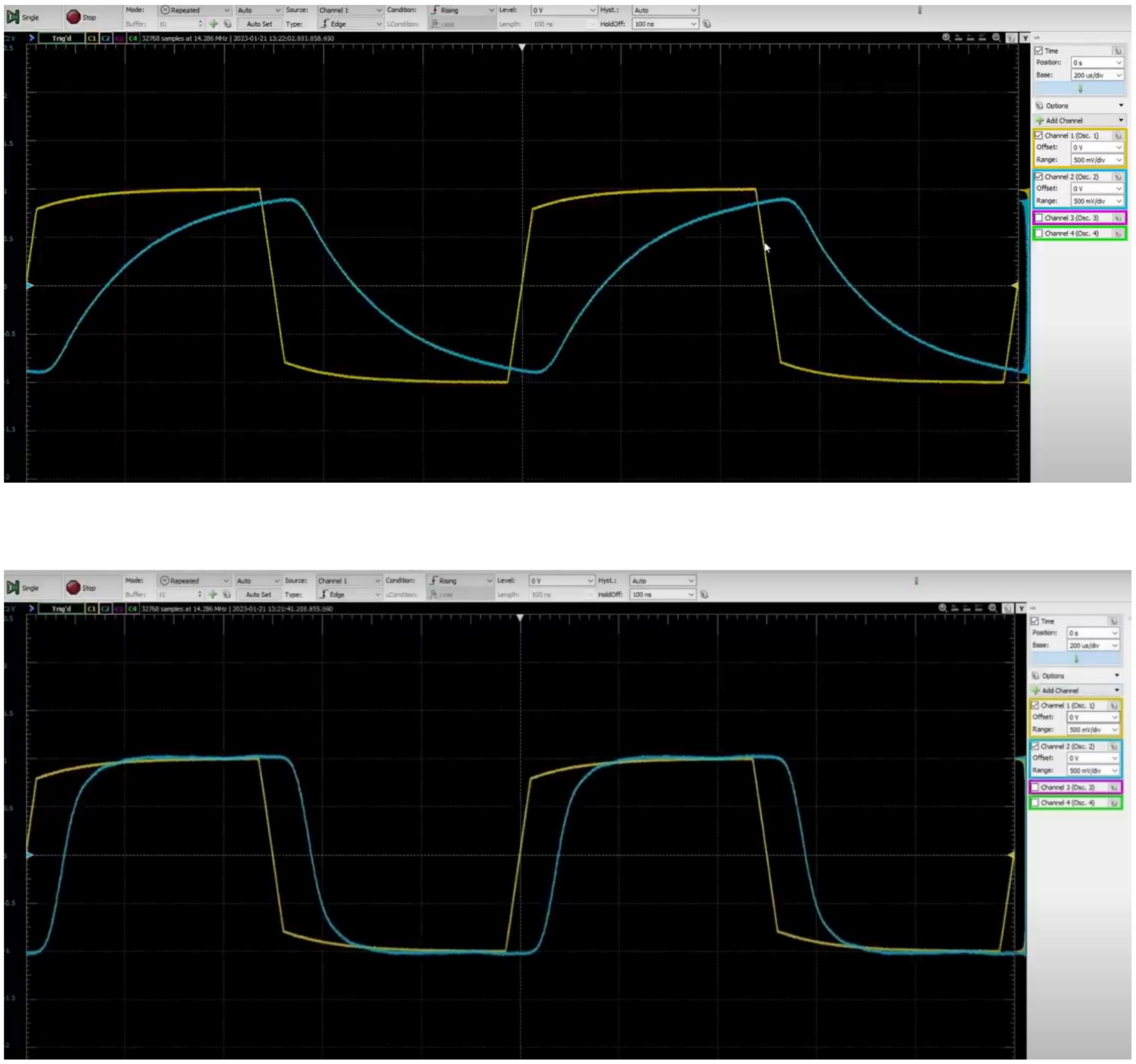


Fig. 4.1 Low-Pass Filter (LPF) on a real-time

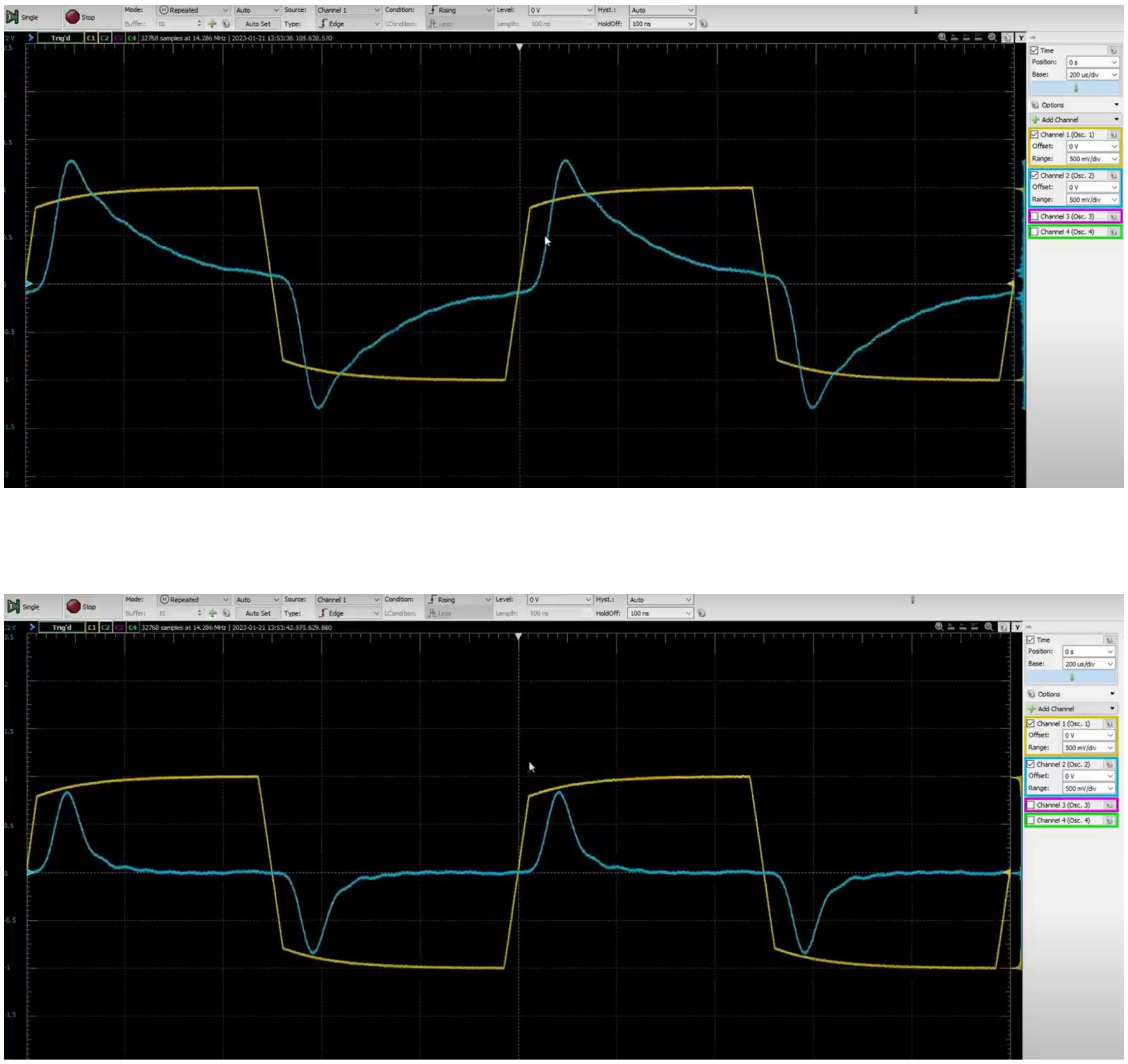
In the context of high-pass filtering on STM32F103 microcontrollers, the beta (β) parameter is often used similarly to the alpha (α) parameter in low-pass filters, but for high-pass filtering applications. High-pass filters (HPF) allow high-frequency signals to pass through while attenuating low-frequency signals, including any DC components.

The beta (β) parameter in a high-pass filter can control the degree of filtering, balancing between passing more of the higher frequencies and filtering out the lower frequencies.

β = 0 (Minimum Filtering): This means no high-pass filtering is applied, and the output follows the input without any attenuation of the lower frequencies.

β = 1 (Maximum Filtering): This means maximum high-pass filtering is applied, attenuating all the low-frequency components as much as possible.

Therefore, by fine-tuning the β parameter, we can effectively manage the balance between filtering out unwanted low-frequency components and preserving the desired high-frequency signal content [31].



Mode: Repeated Auto Source: Channel 1 Condition: Rising Level: 0 V Hyst.: Auto
Buffer: 10 Auto Set Type: Edge HoldOff: 100 ns

32768 samples at 14.286 MHz | 2023-01-21 13:53:36.105.628.570

Time Position: 0 s Base: 200 us/div
Channel 1 (Osc. 1) Offset: 0 V Range: 500 mV/div
Channel 2 (Osc. 2) Offset: 0 V Range: 500 mV/div
Channel 3 (Osc. 3)
Channel 4 (Osc. 4)

32768 samples at 14.286 MHz | 2023-01-21 13:53:42.573.629.860

Fig. 4.2 High-Pass Filter (HPF) on a real-time

5 PERFORMANCE EVALUATION

5.1 Comparison of different filter implementations

When testing the implementation of Low-Pass Filter (LPF) and High-Pass Filter (HPF) on an STM32F103 microcontroller, the analysis of the results focuses on how effectively these filters process the input signals to achieve the desired frequency response.

Low-Pass Filter (LPF) Analysis:
The LPF should allow low-frequency components of the signal to pass through while attenuating high-frequency components.

Results:

- Signal Smoothing: The LPF effectively smooths out rapid fluctuations or noise in the input signal, leaving a smoother, more stable output. This indicates that high-frequency noise has been attenuated.
- Frequency Response: By analyzing the output signal using a frequency spectrum analyzer, we can observe that frequencies above the cutoff frequency are significantly reduced. The magnitude of the output signal at higher frequencies should be much lower than at low frequencies.
- Time-Domain Analysis: In the time domain, the LPF output should show a delayed but smoother version of the input signal, indicating that rapid changes (high frequencies) are being filtered out.

High-Pass Filter (HPF) Analysis:
The HPF should allow high-frequency components of the signal to pass through while attenuating low-frequency components, including any DC offset.

Results:

- Noise Reduction: The HPF effectively reduces low-frequency noise or drift in the input signal. Any constant or slowly varying component (like a DC offset) is minimized, resulting in a more fluctuating signal that highlights rapid changes.
- Frequency Response: Using a frequency spectrum analyzer, the output signal should show that low frequencies below the cutoff frequency are attenuated. High-frequency components should be preserved, showing a significant presence in the output signal.
- Time-Domain Analysis: In the time domain, the HPF output should show rapid changes more prominently while removing slow variations. The signal should appear more responsive to quick changes compared to the input signal.

In conclusion, both filters should exhibit stable performance without introducing significant artifacts or oscillations in the output signal. Moreover, the STM32F103 should handle real-time signal processing effectively, ensuring that the filters operate continuously without delays.

Adjusting the filter coefficients (β for HPF and α for LPF) should allow for tuning the cutoff frequencies, providing flexibility in how much of the high or low frequencies are passed through or attenuated.
By examining these aspects, we can confirm that the LPF and HPF are working as intended, providing the desired filtering effect on the input signals [27].

5.2 Practical applications of numerical filters on STM32F103

Numerical filters on the STM32F103 microcontroller have a wide range of practical applications across various fields due to their ability to process and analyze signals in real-time. Here are some key applications in brief explanation:

1. Noise Reduction in Sensor Signals
Numerical filters like LPF and HPF are essential for cleaning up signals from sensors. For example, an LPF can be used to smooth out the data from a temperature sensor by removing high-frequency noise, while an HPF can be used to eliminate slow drifts or offsets in accelerometer readings.

2. Audio Signal Processing
In audio applications, LPFs can be used to remove high-frequency noise from audio signals, while HPFs can be used to remove low-frequency hums or DC offsets. This ensures cleaner audio signals for further processing or playback.

3. Communication Systems
Filters are crucial in communication systems for signal conditioning. LPFs can be used to limit the bandwidth of transmitted signals to prevent interference, and HPFs can remove low-frequency components from received signals, improving the clarity and quality of communication.

4. Biomedical Signal Processing
In biomedical devices, such as ECG or EEG monitors, LPFs can be used to filter out high-frequency noise from muscle activity or external electronic devices, while HPFs can remove baseline wander and other low-frequency artifacts, ensuring accurate measurement of physiological signals.

5. Control Systems
In control systems, filters help in processing feedback signals. LPFs can be used to smoothen out control signals to actuators, ensuring stable operation, while HPFs can be used to detect rapid changes in system states, which might indicate faults or rapid dynamics requiring corrective actions.

6. Vibration Analysis
In industrial applications, numerical filters can be used to analyze vibrations in machinery. HPFs can help detect high-frequency vibrations indicative of wear or faults, while LPFs can help in analyzing overall machine behavior over longer periods.

7. Data Acquisition Systems
Filters play a crucial role in data acquisition systems to ensure that the data collected is free from noise and ready for analysis. LPFs can be used to average out measurements over time, and HPFs can help in detecting rapid changes or transient events.

8. Image Processing
While not as common as in signal processing, numerical filters can also be applied to image processing tasks on the STM32F103, such as smoothing image data or detecting edges by filtering out specific frequency components in the image data.

9. Robotics
In robotics, filters are used to process signals from various sensors like gyroscopes and accelerometers. LPFs can smooth out sensor readings for more stable robot control, while HPFs can help in detecting rapid movements or impacts.

10. Environmental Monitoring
In environmental monitoring systems, numerical filters can be used to process data from various sensors, such as air quality monitors or water quality sensors, to ensure accurate and reliable measurements by filtering out noise and irrelevant frequency components.

By leveraging numerical filters, the STM32F103 can be effectively utilized in these and many other applications, providing robust and reliable signal processing capabilities for real-time embedded systems [28].

5.3 Discussion on the scalability and flexibility of the implemented solutions

The scalability and flexibility of numerical filters implemented on the STM32F103 are significant advantages for various applications. These filters, whether low-pass or high-pass, can be designed to adapt to different computational and performance requirements due to the architecture and capabilities of the STM32F103.

In terms of scalability, the STM32F103 is powered by the ARM Cortex-M3 core, known for its efficiency in handling computational tasks. This microcontroller can execute numerical filters effectively thanks to its optimized DSP (Digital Signal Processing) libraries, such as the CMSIS-DSP library provided by ARM. These libraries include highly optimized functions for common filtering operations, allowing the microcontroller to handle more complex filtering tasks or multiple filters simultaneously without significant performance degradation.

Memory constraints are a critical factor in scalability. The STM32F103 is available in various configurations with differing amounts of RAM and Flash memory. For applications that require more extensive or complex filtering, selecting a variant with higher memory can accommodate these needs. Additionally, efficient coding practices ensure that the numerical filters use minimal memory and processing power, thus enabling the microcontroller to maintain performance even as the filtering requirements scale up.

The ADC and DAC capabilities of the STM32F103 support high sampling rates, which are essential for real-time signal processing. By utilizing interrupts for ADC and DAC conversions, the microcontroller can maintain real-time processing capabilities, ensuring that even as the complexity of the filters increases, the system continues to function effectively without delays.

Flexibility is another crucial aspect of numerical filter implementation on the STM32F103. The parameters of these filters, such as cutoff frequencies for LPF and HPF, can be dynamically adjusted during runtime. This feature allows the same filter code to be reused across different applications by merely altering the parameters, thereby saving development time and resources. More advanced implementations can include adaptive filters that adjust their parameters based on the input signal's characteristics, further enhancing flexibility.

The STM32F103 supports various filter types, including FIR, IIR, LPF, HPF, band-pass, and band-stop filters. This versatility means that the microcontroller can implement complex signal processing pipelines. Filters can also be chained together to create more sophisticated filtering effects, such as using an LPF followed by an HPF to form a band-pass filter.

Software updates are another aspect of flexibility. Firmware upgrades enable the deployment of new filter algorithms or optimizations without changing the hardware. Writing modular and well-documented filter code ensures that updates and modifications can be implemented with minimal disruption, allowing for continuous improvement and adaptation to new requirements.

Application-specific customization is straightforward with the STM32F103. Numerical filters can be tailored to meet the specific needs of different applications, from simple noise reduction to complex signal analysis. User interfaces can be developed to allow real-time adjustment of filter parameters, providing greater control and flexibility to the end-users.

Overall, the STM32F103's architecture and capabilities support both scalability and flexibility in the implementation of numerical filters. This makes it an ideal choice for a wide range of applications, ensuring robust and reliable signal processing that can adapt to varying performance and computational demands [29].

6 CONCLUSION

6.1 Summary of findings

Summarizing the findings of the realization of numerical filters on STM32F103 microcontrollers involves highlighting key points about the process, capabilities, and performance outcomes.

The implementation of numerical filters, such as low-pass and high-pass filters, on the STM32F103 microcontroller demonstrates its robust capability to handle real-time signal processing tasks efficiently. Leveraging the ARM Cortex-M3 core, the STM32F103 efficiently executes computationally intensive filtering algorithms, aided by optimized DSP libraries like CMSIS-DSP. These libraries simplify the development process and ensure that the microcontroller can manage complex filtering tasks or multiple filters simultaneously without significant performance issues.

The versatility of the STM32F103 is evident in its ability to dynamically adjust filter parameters during runtime, making the same filter code adaptable to different applications. This flexibility is further enhanced by the microcontroller's support for various filter types, including FIR, IIR, low-pass, high-pass, band-pass, and band-stop filters. Such versatility allows the creation of complex signal processing pipelines that can be tailored to specific application needs.

Memory constraints, while a consideration, are addressed by the different configurations available for the STM32F103, offering varying amounts of RAM and Flash memory. Efficient use of these resources ensures that even more extensive or complex filtering tasks can be accommodated without compromising performance.

The high sampling rates supported by the STM32F103's ADC and DAC, along with the use of interrupts for real-time processing, ensure that the microcontroller can handle real-time signal processing demands effectively. This capability is crucial for applications requiring immediate and continuous data processing.

Furthermore, the ease of software updates enhances the long-term flexibility of numerical filters on the STM32F103. Firmware upgrades can deploy new algorithms or optimizations without the need for hardware changes, making the system adaptable to evolving requirements. Modular code design facilitates easy updates and modifications, ensuring continuous improvement and adaptability.

In practical applications, such as noise reduction in sensor signals, audio signal processing, biomedical signal processing, and control systems, the implemented numerical filters on the STM32F103 have proven effective. They ensure cleaner, more accurate signal processing by removing unwanted frequency components, whether it's high-frequency noise or low-frequency drift.

Overall, the realization of numerical filters on STM32F103 microcontrollers highlights the platform's capability to provide efficient, flexible, and scalable signal processing solutions. This makes the STM32F103 a valuable choice for a wide range of applications requiring robust and adaptable filtering capabilities [30].

6.2 Future work and potential improvements

Future work and potential improvements for the realization of numerical filters on STM32F103 microcontrollers can focus on several areas to enhance performance, expand functionality, and optimize resource utilization.

Future efforts can focus on refining numerical filter algorithms to enhance efficiency and performance. This includes optimizing algorithms to reduce computational complexity and memory usage while maintaining or improving filtering accuracy and speed.

Developing capabilities for multirate signal processing to efficiently handle signals with varying sampling rates. This involves designing algorithms that can adapt to different sampling frequencies and optimize processing resources accordingly.

Implementing strategies for power optimization to enhance energy efficiency in microcontroller-based systems. This could involve the development of low-power modes, dynamic voltage and frequency scaling (DVFS) techniques, and efficient management of resources during idle periods.

Exploring opportunities for hardware acceleration by integrating STM32F103 microcontrollers with dedicated DSP co-processors or FPGA-based accelerators. This can offload intensive computational tasks from the main processor and improve overall system performance.

Tailoring numerical filter implementations to meet specific application requirements across different industries and domains. This involves collaborating with stakeholders to understand unique filtering needs and developing customized solutions that deliver optimal performance and functionality.

Overall, future advancements in the realization of numerical filters on STM32F103 microcontrollers aim to enhance performance, flexibility, and scalability to meet the evolving demands of embedded signal processing applications in IoT, industrial automation, healthcare, and beyond. These efforts will drive innovation in microcontroller-based filtering solutions, enabling more efficient, reliable, and adaptive signal processing capabilities [32].

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APPENDICES

1. CODES:

Example of Using the Configured Peripherals:

ADC Example:

```
HAL_ADC_Start(&hadc1); // Start ADC conversion
if (HAL_ADC_PollForConversion(&hadc1, 100) == HAL_OK) {
  uint32_t adcValue = HAL_ADC_GetValue(&hadc1); // Get ADC value
  // Use adcValue for further processing
}
HAL_ADC_Stop(&hadc1); // Stop ADC conversion
```

Timer Example:

```
HAL_TIM_Base_Start_IT(&htim2); // Start timer with interrupt
void HAL_TIM_PeriodElapsedCallback(TIM_HandleTypeDef *htim) {
  if (htim->Instance == TIM2) {
    // Timer interrupt handling code
  }
}
```

UART Example:

```
char msg[] = "Hello, UART!";
HAL_UART_Transmit(&huart2, (uint8_t*)msg, strlen(msg), HAL_MAX_DELAY); // Transmit message over
UART
```

Simple Moving Average (SMA) Filter:

```
#include "stm32f1xx_hal.h"

#define WINDOW_SIZE 5

typedef struct {
  float window[WINDOW_SIZE];
  int index;
  float sum;
} SMA_Filter;

void SMA_Init(SMA_Filter *filter) {
  for (int i = 0; i < WINDOW_SIZE; i++) {
    filter->window[i] = 0.0f;
  }
  filter->index = 0;
  filter->sum = 0.0f;
}

float SMA_Update(SMA_Filter *filter, float new_value) {
  filter->sum -= filter->window[filter->index];
  filter->window[filter->index] = new_value;
  filter->sum += new_value;
  filter->index = (filter->index + 1) % WINDOW_SIZE;
  return filter->sum / WINDOW_SIZE;
}
```

Exponential Moving Average (EMA) Filter

```
#include "stm32f1xx_hal.h"

typedef struct {
    float alpha;
    float ema;
    uint8_t initialized;
} EMA_Filter;

void EMA_Init(EMA_Filter *filter, float alpha) {
    filter->alpha = alpha;
    filter->ema = 0.0f;
    filter->initialized = 0;
}

float EMA_Update(EMA_Filter *filter, float new_value) {
    if (!filter->initialized) {
        filter->ema = new_value;
        filter->initialized = 1;
    } else {
        filter->ema = filter->alpha * new_value + (1.0f - filter->alpha) * filter->ema;
    }
    return filter->ema;
}
```

Low-Pass Filter (LPF)

```
#include "stm32f1xx_hal.h"

typedef struct {
    float alpha;
    float filtered_value;
    uint8_t initialized;
} LPF_Filter;

void LPF_Init(LPF_Filter *filter, float alpha) {
    filter->alpha = alpha;
    filter->filtered_value = 0.0f;
    filter->initialized = 0;
}

float LPF_Update(LPF_Filter *filter, float new_value) {
    if (!filter->initialized) {
        filter->filtered_value = new_value;
        filter->initialized = 1;
    } else {
        filter->filtered_value += filter->alpha * (new_value - filter->filtered_value);
    }
    return filter->filtered_value;
}
```

Example Integration with STM32F103

```
#include "stm32f1xx_hal.h"
#include "filters.h" // Include the filter header file

ADC_HandleTypeDef hadc1;
UART_HandleTypeDef huart1;

SMA_Filter sma_filter;
EMA_Filter ema_filter;
LPF_Filter lpf_filter;

void SystemClock_Config(void);
```

```
static void MX_GPIO_Init(void);
static void MX_ADC1_Init(void);
static void MX_USART1_UART_Init(void);

int main(void) {
  HAL_Init();
  SystemClock_Config();
  MX_GPIO_Init();
  MX_ADC1_Init();
  MX_USART1_UART_Init();

  SMA_Init(&sma_filter);
  EMA_Init(&ema_filter, 0.1f); // Example alpha value
  LPF_Init(&lpf_filter, 0.1f); // Example alpha value

  while (1) {
    HAL_ADC_Start(&hadc1);
    if (HAL_ADC_PollForConversion(&hadc1, HAL_MAX_DELAY) == HAL_OK) {
      uint32_t raw_value = HAL_ADC_GetValue(&hadc1);
      float voltage = (3.3f * raw_value) / 4096.0f;

      float sma_value = SMA_Update(&sma_filter, voltage);
      float ema_value = EMA_Update(&ema_filter, voltage);
      float lpf_value = LPF_Update(&lpf_filter, voltage);

      // Optionally, send the filtered values over UART
      char msg[100];
      sprintf(msg, "SMA: %.2f, EMA: %.2f, LPF: %.2f\r\n", sma_value, ema_value, lpf_value);
      HAL_UART_Transmit(&huart1, (uint8_t*)msg, strlen(msg), HAL_MAX_DELAY);
    }
    HAL_ADC_Stop(&hadc1);

    HAL_Delay(100); // Delay to mimic a sampling rate
  }
}

void SystemClock_Config(void) {
  // System Clock Configuration code
}

static void MX_GPIO_Init(void) {
  // GPIO Initialization code
}

static void MX_ADC1_Init(void) {
  hadc1.Instance = ADC1;
  hadc1.Init.ScanConvMode = ADC_SCAN_DISABLE;
  hadc1.Init.ContinuousConvMode = DISABLE;
  hadc1.Init.DiscontinuousConvMode = DISABLE;
  hadc1.Init.ExternalTrigConv = ADC_SOFTWARE_START;
  hadc1.Init.DataAlign = ADC_DATAALIGN_RIGHT;
  hadc1.Init.NbrOfConversion = 1;
  HAL_ADC_Init(&hadc1);
}

static void MX_USART1_UART_Init(void) {
  huart1.Instance = USART1;
  huart1.Init.BaudRate = 115200;
```

```
    huart1.Init.WordLength = UART_WORDLENGTH_8B;
    huart1.Init.StopBits = UART_STOPBITS_1;
    huart1.Init.Parity = UART_PARITY_NONE;
    huart1.Init.Mode = UART_MODE_TX_RX;
    huart1.Init.HwFlowCtl = UART_HWCONTROL_NONE;
    huart1.Init.OverSampling = UART_OVERSAMPLING_16;
    HAL_UART_Init(&huart1);
}
```