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Real-Time Audio Enchancement: CIC Filter Design for Improved FIR Filter Performance in Digital Signal Processing

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ABSTRACT

The fusion of cascading integrator-comb (CIC) and finite impulse response (FIR) filters to improve audio signals in real time. The development and implementation of an operational audio enhancement system is the main goal. The CIC filter's intrinsic elegance and low computing complexity are utilized for effective decimation or interpolation. Achieving significant rate variations inside the pass band with a flat response to frequency is the key objective. After that, the FIR filter is carefully included to improve performance even more by offering increased frequency shaping and customization to satisfy particular needs for audio improvement. Choosing the right filter order, coefficients, and decimation and interpolated coefficients for the CIC filter, as well as smoothly combining it alongside the FIR filter in a current processing pipeline, are all part of the design stage. The suggested process entails establishing the needs for the system, creating and executing the two filters, and carrying out exhaustive testing and optimization. By providing a thorough method that strikes a balance between computing efficiency, latency, and performance, the work advances the area of immediate fashion sound processing and tackles the difficulties associated with audio improvement in digital systems. The proposed CIC-FIR method achieves a signal-to-noise ratio (SNR) of 12.41 dB, showcasing its performance in maintaining signal fidelity and noise levels. This SNR value serves as a key metric for evaluating the effectiveness of the proposed CIC-FIR approach.

KEYWORDS

Digital Signal Processing; Cascaded-Integrator-Comb Filter; Finite Impulse Response Filter; Latency Optimization

1. INTRODUCTION

A subfield of signal processing known as digital signal processing (DSP) is concerned regarding the tampering significance, and assessment of digital signals. The conversion of analogue data signals to digital signals has several benefits, such as the utilisation of advanced technologies for signal augmentation, selection, enlargement, and modulating [1]. Signals are displayed in DSP as independent value series that are normally captured at predictable times. Sequences like those may reflect a wide range of signals, including sensor data, pictures, audio, and video [2]. Two primary procedures are involved in the conversion of constant signals into distinct forms: normalization and sampling. Selection is the process of recording the on-going signals intensity at periodic times, whereas quantization is the process of giving those intensity discrete numbers to represent them [3]. Digital signal processing (DSP) algorithms are mathematical methods for handling digital signals. Numerous classes, including as filtering, convolution, Fourier analysis, modulation, and statistical signal processing [4], are applicable to these techniques. Finite Impulse Response (FIR) and Infinite Impulse Response (IIR) digital filters are frequently employed for signals improvement, equalisation, and lowering noise. A key DSP approach for quickly calculating a signals discrete Fourier transform (DFT) is the FFT [5].

It makes it possible to analyse data in the range of frequencies domain, which opens up possibilities for modulation, however, audio the compression process, and spectral analysis. The applications for DSP [6] can be found in many different domains, such as biological signal processing, radar and sonar systems, control systems, audio and voice manufacturing, video and image processing, and telecommunication.

DSP is implemented in broadband connections in the field of communications, for instance, to store and retrieve digital data before sending it over transmission paths [7]. The world of audio processing has undergone a revolution in digital signal processing (DSP), which has made it possible to develop cutting-edge methods for improving audio in real time [8]. The creation of effective filters to modify and enhance the properties of sound signals is a critical component of DSP in applications involving sound [9]. The Cascaded Integrator-Comb (CIC) filters has become an efficient tool in such a setting largely because of its ease of use, minimal computational complexity, and compatibility with real-time algorithms [10]. The primary aim of this research is to investigate the possibilities for current time audio improvement offered by CIC filters combined with Finite Impulse Response (FIR) filters [11].

The CIC architectural cascading system of aggregators and combing filters effectively eliminates the high-frequency parts and permits a large drop in speed of data before appreciably altering the signal. Because of this, CIC filters are a good fit for systems like digital audio workplaces, interactive infrastructure

[12], and handheld devices whereby audio improvement is is necessary in instantaneously. Also explore the theoretical basis of CIC filters in this work, as well as how they might be integrated with FIR filters to improve audio quality. Also will investigate design parameters like filter ordering and decimation rate to maximise effectiveness at minimal computing expense. In order to confirm the success rate of the suggested approach in actual situations [13], executions and models will be used combined with analytical evaluations. The results of the current investigation have the potential to improve real-time audio enhancing modern facilities by offering an improved computationally productive approach that satisfies the requirements of contemporary musical processing programmes. The results may make an impact on a number of sectors, such as professional sound manufacture, electronics for consumers, and internet access, whereby excellent quality, low-latency technology signal processing is crucial.

The key contributions of this study are as follows:

1.1Efficient Rate Change: The main contribution of the CIC filter is how well it implements changes in sample rate using comb and cascaded integrator stages. The CIC filter achieves considerable rate variations with a very basic structure, minimising computing complexity and resource needs. This is achieved by taking advantage of the intrinsic features of the comb sections for decimation and interpolation and the integrator stages for filtering.

1.2 Flat Frequency Response***:*** The CIC filter's flexibility to deliver a flat frequency spectrum inside the pass band is one among its primary contributions. This is especially helpful in applications where signal integrity and distortion reduction are critical, including digital messaging devices and audio processing, where a constant amplitude response over the applicable frequency bandwidth is essential.

1.3 Real-time Digital Signal Processing***:*** The application of the CIC filter in actual time processed digital signals is improved by its combination with a Finite Impulse Response (FIR) filter. Applications like audio enhancements can benefit from the combined CIC-FIR filter system's versatile and efficient technique for accomplishing both extra spectrum shaping and rate change. The construction of effective and adaptable computational pathways that satisfy particular needs in a range of digital data transfer and audio processing platforms is made possible by this work.

The format for the enduring paragraphs is as follows: The relevant work based on various methodologies for diabetes prediction is examined in section II, and the research gap is identified in section III. The proposed framework is explained in the Section IV. The outcomes and considerations are covered in Section V; the prospective applications for the future are covered in Section VI.

2. RELATED WORKS

Micro-Electro-Mechanical Systems (MEMS) sensor arrays are becoming widely used in microphone sources identification due to their minimal consumption of electricity and ability to operate in immediate time circumstances. The goal of this research project is to achieve stringent energy conservation standards while precisely establishing the Direction-Of-Arrival (DoA) [14] of sound sources. In order to build adaptable systems, the talk examines architectural implications of crucial procedures for sound placement. A range of effectiveness methods that correspond all the unique features of the energy-effective design that is being described are suggested and assessed.

Finite Impulse Response (FIR) and Infinite Impulse Response (IIR) filters are essential components in the vast field of Digital Signal Processing (DSP) because they allow for the layout of intricate signal processing structures for uses in flexible antenna approach health care signal analysis, radar signal processing, and automated operation [15]. The Parks-McClellan method is used to analyse produced at random input that exists in Random Access Memory (RAM) and store the results in Read-Only Memory (ROM). Evaluation criteria for the presented AM-CSA IIR (Array Multiplier-Carry Skip Adder Infinite Impulse Response) filter include slices, delay, proportional delay, flip flops (FF), Look-Up Table (LUT), and the level of output. By targeting important characteristics that are crucial for evaluating efficacy in real-world DSP programmes, this research helps to optimise technology effectiveness in IIR filter architectures. Although the present design seeks to maximise hardware utilisation, the search for an alternative optimal filtration on FPGA technology is still necessary to overcome these obstacles, guaranteeing continuous attempts to reduce power consumption and achieve better quantum delay effectiveness. The search for different layouts is an example of the constant requirement for development to keep up with the changing needs of DSP implementations.

The need for optimised alternatives has increased in today's multimedia signal processing environment, especially regarding communication and current signal processing systems [16]. This work explores the architecture and application of infinite and finite impulse response filters, which are essential parts of sophisticated data processing algorithms. This research delves into several types of frequency domain reflection (FIR) filters: traditional moving average (MA) FIR filters, fast MA FIR filters using look-ahead arithmetic, typical IIR filters using the integrator-comb section (CIC) approach, and fast IIR filters using look-ahead computation. The Quartus II 13.1 synthesis tool is utilised to assess and compare the effectiveness measures of regular and fast MA FIR and IIR filters, covering Logic Elements (LEs), speed, and energy use, using an Altera EP4CE115F29C7 field-programmable gate array (FPGA) controller. The findings demonstrate the suitability the suggested FIR and IIR filters are for VLSI signal analysis workloads due to their efficient architecture, which places a strong emphasis on speed. The complete implementation of various methods for filtering into FPGA-based signal processing architectures will require resolving these possible restrictions as DSP implementations develop.

Considering an emphasis on interference reducing, this research evaluates filtering designed for traditional overestimated audio wave shaping systems. This research looks into wave shaping sinusoidal structures by hard clipping and observes that higher anti-aliasing distortions and spectral enhancement are caused by interruptions in the asymmetric transference function [17]. The results show that interpolation of linearity equalized by an extensive frequency filter is a good enough further sampling technique. Interpolated FIR, elliptic, and cascaded integrator-comb filters all show benefits above elementary situations in the setting of decimate. Interestingly, when combined by the linearly interpolator, the cascaded integrator-comb filter is the only decimated filtering that has been evaluated and found to provide visually adequate interference suppressing over the whole bandwidth range. There is still work to be done on the topics of determining the best filter sequence, from fine-tuning parameters, and comparing decimated filtering. As a result, the thorough development of these parts is postponed until later, highlighting the necessity of continued study to improve and optimise both the decimation and up sampling procedures in the goal of achieving greater effectiveness in sound applications that utilise signal processing.

With a focus on simple construction via multiplier-less building design, this study discusses the creation of a computerized Finite Impulse Response (FIR) filter for computerised hearing assistance systems [18]. The emphasis is on the decimated filter's critical function in accomplishing this objective. The suggested design aims to minimise energy usage and storage compared to previous methods by introducing estimated 4:2 compressing arithmetic in a memory-less Differential Adder (DA) based FIR filter framework. The designs, which was created employing Synopsys is Applications Specialised Combined Circuit (ASPIC) generator and synthesised on 90 nm technological devices, shows a 10% enhancement over OBC DA structure and a decrease of 45% in area latency product when contrasted with cyclic paradigm. FPGA implementations validates lesser segments compared to current concepts, demonstrating the suggested the architectural effectiveness. This limitation highlights the possibility for more study to increase the adaptability and efficiency of disintegration filtration in the field of audiology innovation.

This study proposes a pure digital structure that does away with the requirement for analog-to-digital (AD) and digital-to-analog (DA) conversion devices, thereby introducing an innovative method to active noise control (ANC) systems [19]. This invention reduces the demand for expensive and bulky AD and DA converters while also streamlining the entire system design. A prospective fundamental change in the architecture of ANC systems is presented by combining a set of electronic acoustical devices with an FPGA-based temporal framework. Such a combination offers significant benefits in the areas of simplicity and profitability while preserving efficient noise suppression. Although the real computerised ANC method that has been suggested is a positive architectural change, it may have drawbacks. The use of FPGA-based technology and digital acoustical components may provide difficulties for the system to scale up and flexibility to a range of noise situations. Furthermore, more testing is required to determine if real-time supplementary route modelling is practically feasible under the suggested cardiac order, and particular focus should be given to the benefits and drawbacks involved in doing away with AD and DA conversions. To overcome these shortcomings and guarantee the resilience and suitability of the suggested computerised ANC structure for a range of actual-life situations, more study and testing are necessary. The study [20] simulates decimator architectures using Simulink and highlights how these arrangements affect measurement parameters. The analysis shows that using a single-stage decimator requires more storage capacity, a more advanced filter, and longer simulated times. This study emphasises the trade-offs involving filter sequence, from storage needs, and computing efficiency when attempting to achieve the needed spectrum lowering, and it provides substantial insight into the optimisation of decimator topologies for WLAN systems. Despite its constraints, the investigation offers useful insights into the performance and structure of multi-rate decimators for WLAN systems. The study might not prove immediately useful to alternative WLAN specifications or a variety of converting frequencies needs because it is exclusive to WLAN-b operations. Furthermore, the research leaves an opportunity to conduct further investigation of sacrifices combining computational demands and rapid processing limitations, as it emphasises largely on efficacy indicators like storage components. For an even more sophisticated knowledge of multi-rate signal handling in WLAN innovations, a subsequent investigation might broaden its reach to include a greater variety of WLAN specifications and consider a more complete set of outcomes metrics.

3. PROBLEM STATEMENTS

The investigation offered covers a wide range of issues related to signal being processed, including advanced audio wave shaping systems, digital FIR filters for audiological programmes, fresh ideas related for active disturbances management, optimisation of FIR and IIR filters, MEMS sensor sets for microphone designed source proof of identity, and multi-rate signal manipulation for WLAN advances in technology [20]. Although each investigation makes a significant contribution to the discipline, the body of work as a whole draws attention to a number of issues and problems. First of all, the implementation of FPGAs in MEMS sensor collections for audio source identification is beneficial for complicated circumstances [19]. However, additional analysis is needed to determine possible drawbacks in real-world uses, particularly with regard to both electrical effectiveness and computing needs. Even though FIR and IIR filter optimisation has shown increases in CPU utilisation and system efficacy, problems with power usage and proportionality latency may arise in real-world scenarios. Furthermore, the suggested genuine digital ANC technology shows promising but has to be validated in various noisy settings and its capacity to grow evaluated. Primarily concentrating on WLAN-b programmes, the research on multi-rate signal handling for WLAN techniques raises questions about its generalizability to various other WLAN protocols and the investigation of metrics of performance transcend the components of storage. While introducing an interesting method, the analysis on computerised FIR filters used for hearing device systems also acknowledges possible limits in energy usage and proportionality delay, indicating the need for more research to solve these issues. In conclusion, these studies make a substantial contribution to their appropriate fields, but taken as a whole, they highlight the significance of tackling issues like energy efficiency, scalability in the real world, and thorough evaluations of function in order to increase the practical value of those advances in processing signals. Subsequent investigations ought to focus on addressing these obstacles in order to augment the efficiency and adaptability of these gadgets across a range of uses.

4. INTEGRATED FRAMEWORKS FOR DIGITAL SIGNAL PROCESSING

This paper provides an in-depth analysis of real-time audio improvement achieved by combining a finite impulse response (FIR) filter with a cascaded integrator-comb (CIC) filter in digital signal processing. In order to improve audio shaping, the study focuses on employing the CIC filter in a seamless manner with the FIR filter to achieve flat frequency response and effective rate shifts. Testing, system integration, and filter design are all included in the suggested procedure. The study offers a diverse solution with potential applications in digital communication systems and audio processing. Its contributions arise from its balanced consideration of computing efficiency, latency, and performance. Adaptive approaches and hardware optimisation may be used in future work to further improve the system.

4.1 CIC Filter

One form of finite impulse response (FIR) filter that is frequently utilised for decimation and interpolation in processing of digital signals usage, particularly in the areas involving digital audio restoration and communications, is the cascaded integrator-comb (CIC) filter [21]. The cascaded integrator and comb stages that make up the CIC filtering structure function as low-pass filters for the integrators and as decimators or interpolators for the comb stage sections. The main goal of the CIC filter is to effectively apply modifications to the rate of sampling by taking use of the built-in interpolation and decimation capabilities of its framework. CIC filters are renowned for their ease of use, minimal computing overhead, and capacity to handle significant rate variations with efficiency and a flat pass band spectrum [22]. They are widely used in actual time signal processing solutions because they find usage in a variety of signal processing systems, such as digital receivers, broadband connections, and audio decoding chains.

Steps to design Cascading Integrator-Comb (CIC) filter:

1) Determine Decimation and Interpolation Factors: The interpolation factor (R) and decimation factor (M) are two essential factors that determine how the CIC filter behaves. They are associated with the total change in filter rate. The connection is made possible by (1):

$$L = M * R \tag{1}$$

where, L is the factor of total rate change, M represents the decimation factor and the interpolation factor is denoted by R. L might be adjusted for audio programmes according to the required enlargement or decrease of frequency.

2) Filter Order and Stages: The integrator and comb stages make up the CIC filter. N, the number of steps, is derived from the interpolation and decimation factors as expressed in (2):

$$N = \frac{L-1}{3} \tag{2}$$

The number of phases may be reasonably estimated using this formula. The precise figure might need to be changed depending on the demands of your design.

3) Comb Filter Coefficients: Decimation or interpolation is done by means of the comb filter. By simplifying and extending this transfer function, the coefficients for the comb filter may be obtained. The following provides the comb filter transfer function is given in (3):

$$H_{Comb}(z) = \frac{1-\frac{1}{z^M}}{1-\frac{1}{z}} \tag{3}$$

4) Integrator Filter Coefficients:

Identical to the comb filter, the integrator filtering's coefficients may be determined. High-frequency noise has to be filtered out using the integrator filter. The integrator's transfer equation is described in (4):

$$H_{int}(z) = \frac{\left[1-\frac{1}{z}\right]^N}{\left[1-\frac{1}{z^R}\right]^N} \tag{5}$$

5) CIC Transfer Function: The integrator and comb transfer functions can be cascaded to obtain the overall transfer function of the CIC filter as expressed in (6):

$$H_z = H_{comb}(z) * H_{int}(z) \tag{6}$$

4.2 FIR Filter

A computational filter type known as a limited Impulse Response (FIR) filter [23] is distinguished by an impulse response that has a limited duration. Because FIR filters don't contain feedback loops, they are naturally stable and possess linear phasing features, ranging in contrast to Infinite Impulse Response (IIR) filters. A limited set of parameters that establish the averaged contributions from previous input values to the present output define the impulse response of a FIR filter. Because FIR filters may be precisely designed, they are frequently employed in a broad range of programmes, such as interactions, audio equalisation, and signal analysis. When designing a FIR filter, variables including cutoff frequency bands, filter order, as well as desired behaviour must be specified. Windowing and frequency sampled are often used techniques in this process [24]. In situations where a consistent and manageable frequency response is essential, FIR filters are a good choice due to its simplicity, stability, and capacity to produce phase stability. Defining the demands and then building the filter in accordance with them are the steps involved in creating a Finite Impulse Response (FIR) filter for audio enhancements.

Steps to design Finite Impulse Response (FIR) Filter:

1) Specify Requirements: The needs for the filters must be specified until the FIR filter can be designed. Included as well is the intended frequency response, which may be described by variables like transition bandwidth, stop band attenuation, pass band ripple, and filter type (e.g., low-pass, high-pass, band-pass).

2) Filter Order: One important factor that controls the total number of connections in the FIR filter is the filter order N. It has a direct bearing on how sharply the filter responds to frequencies. Either empirical principles or filter designing instruments are frequently used to establish the filter order.

3) Frequency Sampling: The intended response of frequencies is specified at discrete frequencies points using the spectral sampling approach is expressed as in (7). Normalised inclined frequencies (ω) between 0 and π (or 0 to 1 in normalised frequency) are frequently used for applications that require audio.

$$H_d(e^{j\omega}) = \sum_{k=0}^{N} \blacksquare (k) \frac{1}{e^{j\omega k}} \tag{7}$$

where, the intended frequency response is $H_d(e^{j\omega})$ and the filter coefficients are $h(k)$.

4) Window Function: The windowing function is frequently used to the idealised impulse response in order to transform a frequency-domain sampling arrangement into a realisable FIR filter. Window functions like Hanning, Blackman, and Hamming are often used. Equation (8) gives the windowed impulse response, $h_w[k]$.

$$h_w[k] = h[k] * w[k] \qquad (8)$$

where, the windowed responses to impulses is $h_w[k]$ and the selected window function is $w[k]$.

5) Filter Coefficients: The windowed impulse response is subjected to the inverse discrete Fourier transform (IDFT), which yields the final set of filter coefficients as given in (9):

$$h[k] = \frac{1}{N}\sum_{n=0}^{N-1} h_w[n]e^{j\frac{2\pi kn}{N}} \qquad (9)$$

4.3 Integration of CIC- FIR Filter

By combining these two filtration types, one may integrate a Finite Impulse Response (FIR) filter with a Cascaded Integrator-Comb (CIC) filter to obtain improved signal analysis in digital circuits. Usually used for interpolation or decimation, the CIC filter effectively modifies the sample rate of the signal. The FIR filter, which is intended to offer further spectrum structuring or other desired qualities, comes next. The FIR filter receives its input from the CIC filter's output during the integration phase. Applying the transferring functions of the separate filters yields the whole transfer function of the entire system. Effective control over the frequency responsiveness and the rate at which changes occur of the signal being sampled is made possible by this cascaded technique. Real-time digital signal extraction uses an integrated CIC-FIR filter system, especially in audio enhancing circumstances whereby the cascaded filters cooperate to produce the appropriate screening and rate-changing impacts. To attain the intended audio improvement, cascading the CIC and FIR filters is a necessary step in combining them into an instantaneous noise reduction pipeline. Applying the transfer functions of the different filters yields the whole transfer parameter for the cascaded system (H_{total}).

Cascading Transfer Functions: $H_{CIC}(z)$ represents the transfer function of the CIC filter, and $H_{FIR}(z)$ represents the transfer function of the FIR filter. The following represents the entire transfer function $H_{total}(z)$ is given in (10):

$$H_{total}(z) = H_{CIC}(z) * H_{FIR}(z) \qquad (10)$$

Equation (10) serves as the cascaded system's overall transfer function.

Overall Frequency Response: By assessing the transfer function at various frequency bands, one may determine the complete frequency response $H_{total}(e^{j\omega})$. For instance, the frequency response at a given frequency ω_k may be obtained in discrete-time systems as follows in (11):

$$H_{total}(e^{j\omega_k}) = H_{CIC}(e^{j\omega_k}) * H_{FIR}(e^{j\omega_k}) \qquad (11)$$

Real-Time Implementation: Usually, filtering algorithms are utilised in a streaming form in order of application in a real-time processing of audio system. The CIC filter processes the input audio signal $x[n]$, and the output is then sent through the FIR filter. The CIC output is convolved with the FIR filter coefficients to get the general result $y[n]$ is given in (12):

$$y[n] = x[n] * h_{CIC}[n] * h_{FIR}[n] \qquad (12)$$

where, the sound wave that is being input is represented by $x[n]$, the CIC filter's impulses response is expressed as $h_{CIC}[n]$, the FIR filter's impulses response is represented by the expression $h_{FIR}[n]$ and the combining convolution is indicated by '$*$'.

5. RESULTS & DICUSSIONS

5.1 Testing

1) Frequency Response Analysis: To examine the frequency response of the amplified audio stream, use frequency analysis (FFT) software as mentioned in Equation (13):

$$Y(e^{j\omega}) = X(e^{j\omega}) * H_{total}(e^{j\omega}) \qquad (13)$$

where, $Y(e^{j\omega})$ is the frequency response of the output, $X(e^{j\omega})$ is the frequency response of the input and $H_{total}(e^{j\omega})$ is the overall transfer function.

A CIC filter's frequency response study is crucial for determining its performance attributes and applicability for particular signal processing applications. With the use of this analysis, the design parameters may be optimised to provide the intended frequency domain filtering effects. Fig. 1 represents the frequency response analysis [25] of CIC filter.

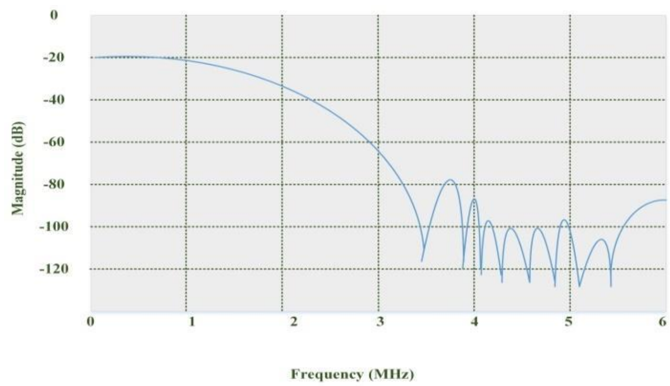

Fig. 1. Frequency Response Analysis of CIC

Filter A Finite Impulse Response (FIR) filter's frequency response analysis is essential to comprehending its behaviour in the frequency domain. The capacity to sculpt a signal's spectrum using the impulse response coefficients of the signal is what defines the frequency response of the FIR filter. Three important factors are the transition band width, stopband attenuation, and passband ripple. While stopband attenuation gauges the filter's capacity to suppress frequencies outside of its passband, passband ripple depicts amplitude changes inside the filter's passband. The range between the passband and stopband is defined by the transition band width, which affects how rapidly the filter moves between these areas and optimising FIR filter designs for particular applications and Fig. 2 represents the frequency response analysis of FIR filter.

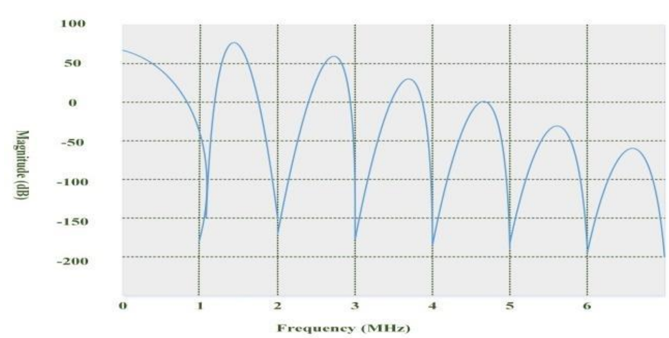

Fig. 2. Frequency Response Analysis of FIR Filter

2) Impulse Response Analysis: Equation (14) examines the platform's impulse response to make sure it offers the required time-domain features.

$$y[n] = x * h_{CIC} * h_{FIR} \quad (14)$$

3) Signal-to-Noise Ratio (SNR) Measurement: To make sure the appliance isn't adding too much noise, check the SNR of the improved audio stream as expressed in Equation (15).

$$SNR = 10 \cdot log_{10}\left(\frac{signal\ power}{noise\ power}\right) \quad (15)$$

methods	SNR (dB)
Sigma-Delta [1]	113
FPGA [2]	16.8008
Proposed CIC-FIR	12.41

Table I. Comparative Analysis of Signal to Noise Ratio

The table I depicts the comparative analysis of Signal-to-Noise Ratio (SNR) performance of different methods is compared based on their respective dB values. The Sigma-Delta method achieves an SNR of 113 dB, indicating its high signal fidelity. In contrast, the FPGA method exhibits a lower SNR of 16.8008 dB. A proposed CIC-FIR method presents an intermediate SNR value of 12.41 dB, suggesting a trade-off between the Sigma-Delta and FPGA approaches in terms of signal quality. The graphical representation of the comparative analysis of Signal-to-Noise Ratio (SNR is displayed in Fig. 3.

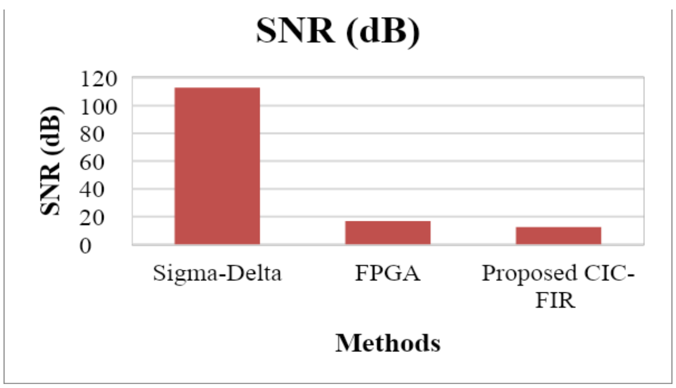

Fig. 3. Comparative Analysis of SNR (dB)

5.2 Optimization

5.2.1 Latency Optimization: Equation (16) reduces latency by minimizing the delay in signal processing.

$$Latency = \frac{Number\ of\ Samples\ Processed}{Sampling\ Rate} \quad (16)$$

5.2.2 Computational Efficiency: To guarantee processing in real time, make sure the methods and code are optimised for computational performance with Equation (17).

$$Computational\ Efficiency = \frac{Useful\ Work\ Done}{Total\ Computational\ Work} \quad (17)$$

5.2.3 Power Consumption Optimization: Equation (18) reduces power consumption by optimising the hardware design and algorithms, which is important for battery-powered devices.

$$Power\ Consumption = Voltage(V) * Current(I) \quad (18)$$

The table 1 displays information on the milliwatt-hour (mW) power dissipation of finite impulse response (FIR) filters with different tap lengths. The type of FIR filter is indicated in the "Filters" column along with the tap lengths for each filter, which correspond to the number of coefficients in the filter. The amount of electricity that each FIR filter dissipates while in use is shown in the "Power Dissipation (mW)" column. The 16-tap FIR filter has a marginally lower power consumption of 133 mW compared to the 140 mW of the 8-tap filter. At 140 mW, the power dissipation of the 32-tap FIR filter is equal to that of the 8-tap filter. Finally, there is a minor increase in power dissipation to 145 mW for the 64-tap FIR filter.

Table II. Power Dissipation of FIR

FIR	Power Dissipation (mW)
8-tap	140
16-tap	133
32-tap	140
64-tap	145

The power consumption characteristics associated with various tap lengths in FIR filters are highlighted in this material, which is essential for engineers and designers. According to the statistics, the 16-tap filter achieves a comparatively lower power dissipation than its equivalents, and the power dissipation increases moderately with longer tap lengths. The graphical representation of the FIR's Power Dissipation is displayed in Fig. 4

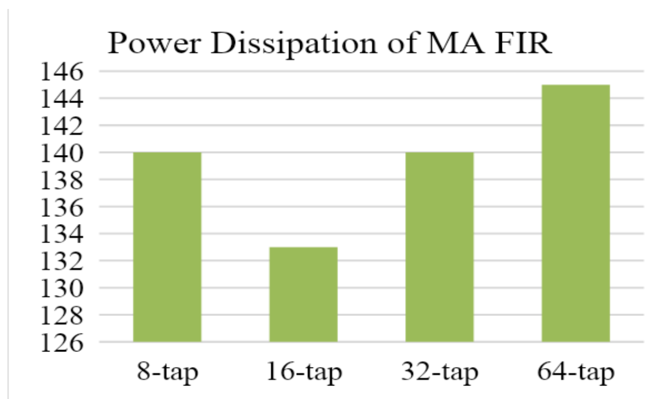

Fig. 4. A Graphical Depiction of the MA FIR's Power Dissipation

5.3 Logic Element for the FIR

The kind of moving average (MA) filters—more especially, their tap lengths—are shown in this column. The number of weights or coefficients utilised in the filter is referred to as the tap length. The performance of each MA filter is shown in this column as a function of frequency, expressed in megahertz (MHz). It displays the rate at which input data may be processed by the filters.

Table III. Applying Logical Components to the FIR

FIR	Performance (MHz)
8-tap	170
16-tap	175
32-tap	179
64-tap	180

The efficiency features, expressed in megahertz (MHz), of Moving Average (MA) filters with different tap lengths are displayed in the table 2. The look at four different types of MA filters with tap lengths of 8, 16, 32, and 64. The rate at which each filter can process input data is shown in the "Performance (MHz)" column. The performance of the 8-tap MA filter is 170 MHz; the 16-tap filter performs slightly better at 175 MHz; the 32-tap filter performs even better at 179 MHz; and the 64-tap filter performs best at 180 MHz. This highlights the trade-off between computational resources and enhanced filtering features in MA filters by suggesting a trend where longer tap lengths correlate with faster processing speeds. The graphical representation of the FIR's logic parts is displayed in Fig. 5.

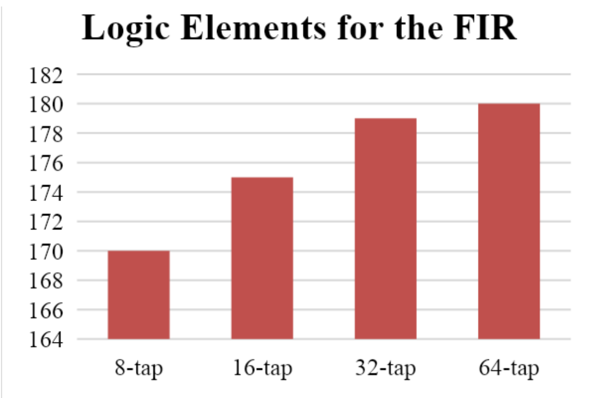

Fig. 5. A Graphical Depiction of the FIR's Logic Components

6. CONCLUSION

The reliable and effective method for digital signal processing is the combination of a Finite Impulse Response (FIR) filter with a Cascaded Integrator-Comb (CIC) filter for real-time audio improvement. A flexible approach to handling the intricacies of audio processing is provided by the cooperative synergy of the FIR filter for further frequency shaping and the CIC filter for rate change. This integrated system exhibits remarkable potential for applications in digital communication systems and audio processing because it strikes a balance between computing efficiency, latency, and performance. The study presents a thorough workflow that includes filter design, system integration, and testing. This workflow advances real-time signal processing methodologies and serves as a valuable basis for future research and development in the fields of digital signal processing and audio enhancement. To further improve the system's flexibility, future research might investigate adaptive strategies for dynamically changing filter settings in response to changing audio characteristics. Furthermore, the practical usefulness of the combined CIC-FIR filter system in various real-time audio processing scenarios will be enhanced by exploring hardware optimization ways to implement it on resource-constrained systems

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