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A Novel Approach for Optimal Design of Sample Rate Conversion Filter Using Linear Optimization Technique

DIVYA GAUTAM, KAVITA KHARE, AND BHAVANA P. SHRIVASTAVA, (Fellow, IEEE)
Department of Electronics and Communication Engineering, Maulana Azad National Institute of Technology, Bhopal 462003, India
Corresponding author: Divya Gautam (divya.gautam7@gmail.com)

ABSTRACT This paper presents a sample rate conversion filter for decimation with flat passband. The proposed linear programming optimization (LPO) technique improves the magnitude response of filter with least computational complexity. Computational complexity has been a major factor in selection of decimation filter. Simulation results indicate that the proposed filter shows passband droop less than 0.007 dB with 50.5% decrease in computational complexity. The proposed filter eliminates the need of compensator.

INDEX TERMS Sample rate conversion, FIR filter, optimization, passband.

I. INTRODUCTION

Multi-rate signal processing has emerged as an essential requirement in the domain of digital signal processing. It requires resampling the original signal at a different sample rate [1]–[3]. The desired sampling rate is achieved by either increasing or decreasing the original sampling rate without destroying the signal components. Decimator decreases the original sampling rate and interpolator increases the sampling rate. Such changes in the sampling rate demand proper filtering techniques to conserve original signal components.

Comb is the simplest multiplierless decimation filter. However, its magnitude response exhibits a considerable droop in the passband and low attenuation in the stopband. The transfer function of the comb filter is given by:

$$H^K(z) = \left(\frac{1}{M}\frac{1-z^{-M}}{1-z^{-1}}\right)^K = \left[\frac{1}{M}\sum_{k=0}^{M-1} z^{-k}\right]^K \quad (1)$$

where M is the decimation factor and K represents the number of stages.

An efficient architecture of comb decimation filter was proposed by Hogenauer in the year 1981, popularly termed as CIC (Cascaded Integrated Comb) decimation filter [4] as shown in Fig. 1. It consists of two main sections: an integrator section and a differentiator section, separated by a down-sampler. It moves part of the filtering at lower rate. As all the filter coefficients are unity, the hardware implementation of this filter is highly economical. In 1997, Tapio Saramäki and Tapani Ritoniemi proposed another comb filter structure for decimation, popularly known as the Saramäki-Ritoniemi structure [5]. It uses the sharpening polynomial as tapped interconnection to improve the frequency response of decimation filter.

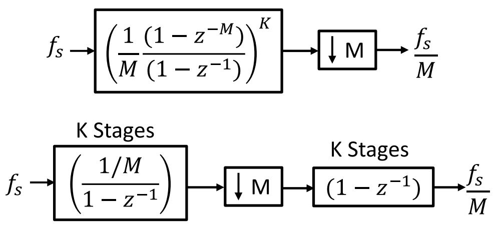

FIGURE 1. CIC structure by hogenauer.

Research in the design of a comb decimation filter are focused on:

1) Minimization of the passband droop
2) Increase alias rejection (stopband attenuation)
3) Improve magnitude response with least increase in computational complexity

Developing strategies to overcome the above limitations has been an active area of research. To improve magnitude characteristics comb based zero rotation technique was introduced by Presti [6]. However, it suffers from two drawbacks; introduction of multiplier in structure and susceptible to imperfect pole-zero cancellation. The problem is solved by using nonrecursive filter in polyphase form [7]. These approaches provide solutions, but the passband improvement cannot be completely controlled. Therefore, these methods are convenient for narrow bandwidth of interest.

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The passband droop and alias rejection can be improved by employing sharpening techniques like Kaiser-Hamming sharpening [8]–[12], Chebyshev polynomial sharpening [13], [14]. Sharpening techniques are effective and does not suffer from finite-precision effects as in comb based zero rotation method. Further to reduce the computational complexity the overall decimation factor M is split as $M = M_1M_2$ where both $M_1, M_2 \in \mathbb{Z}^+$, popularly known as two-stage decimation. Sometimes these two-stage decimation structures use compensator to improve the overall magnitude response of the filter [7], [13], [15]. Later compensators are designed for sharpened CIC. The sharpening coefficients and compensator's coefficients are expressed as SOPOT (Sum Of Power Of Two), converting multipliers to shift and add network. This facilitates a multiplierless implementation [16]–[18].

II. METHODS

Aim is to design a comb decimation filter with minimum passband droop while providing considerable attenuation in stopband. In this paper we propose an approach for comb decimation filter sharpening. Designing of filter and calculations to find sharpening coefficients are done in MATLAB. The optimized framework to design sharpened comb-based filters to attain minimum passband distortion is provided. The optimized sharpening coefficients are in SOPOT (Sum Of Power Of Two) form leading to multiplierless structure, which is important for low power applications.

The rest of this paper is organized as follows. Section III details the implementation of the proposed decimation filter. Section IV compares the magnitude response of the proposed filter with existing ones. Finally, Section V concludes this paper.

III. PROPOSED COMB DECIMATION FILTER

We propose a cascade of comb filter followed by linearly sharpened structure to realize a decimation filter with improved magnitude characteristics. Second stage sharpening polynomial is derived using optimization technique. The overall decimation factor M is split as $M = M_1M_2$. The design parameters include M_1, M_2, Q, N, η_k. The transfer function of the proposed comb filter is given by:

$$H_{SH}(z) = H_{11}(z)H_{22}(z^{M_1}) \tag{2}$$

where,

$$H_{11}(z) = \left[\frac{1}{M_1}\sum_{k=0}^{M_1-1} z^{-k}\right]^Q = \left[\frac{1}{M_1}\frac{1-z^{-M_1}}{1-z^{-1}}\right]^Q \tag{3}$$

$$H_{22}(z) = \sum_{k=0}^{N} \eta_k z^{-(D_k)} H_2^k(z) \tag{4}$$

Here,

$$H_2(z) = \left[\frac{1}{M_2}\frac{1-z^{-M_2}}{1-z^{-1}}\right]^Q ; D_k = \frac{(M_2-1)}{2}(N-k)Q \tag{5}$$

η_k and D_k are sharpening coefficient and delay respectively. N is the degree of sharpening.

Substituting (4) in (2), we get:

$$H_{SH}(z) = [H_{11}(z)]\left[\sum_{k=0}^{N}\eta_k\left[H_2(z^{M_1})\right]^k z^{-M_1(D_k)}\right] \tag{6}$$

The magnitude response of the above transfer function is given as:

$$\left|H_{SH}(e^{j\omega})\right| = \left|\frac{1}{M_1}\frac{\sin\frac{\omega M_1}{2}}{\sin\frac{\omega}{2}}\right|^Q \left|\sum_{k=0}^{L}\eta_k\left[\frac{1}{M_2}\frac{\sin\frac{\omega M}{2}}{\sin\frac{\omega M_1}{2}}\right]^{Qk}\right| \tag{7}$$

As seen from the above equation the magnitude response of the proposed two-stage comb filter is the multiplication of two individual magnitude responses; where the first one corresponds to the magnitude response of the comb filter and second to sharpened structure which depends on the coefficients η_k.

A. LINEAR PROGRAMMING OPTIMIZATION

Linear programming optimization is the problem of finding a vector x that minimizes a linear function $f^T x$ subject to linear constraints.

$$\min_x f^T x \quad \text{such that} \begin{cases} A\cdot & x \le b \\ A_{eq}\cdot & x = b_{eq} \end{cases} \tag{8}$$

There are three algorithms to solve this problem.

1) Dual-simplex algorithm
2) Interior-point algorithm
3) Interior-point-legacy algorithm

The general steps for these algorithms are same. Presolve or preprocessing, meaning simplification and conversion of the problem to a standard form. Then generate an initial point. Finally, the iterations to solve the equations [19].

By default, dual-simplex algorithm is used to find optimal solution for linear programming. It is recommended due to least memory usage and fast processing. There can be potential inaccuracy with interior-point algorithm and interior-point-legacy algorithm can be slower, less robust, or use more memory [20]. Fig.2 shows flow chart of LPO using dual-simplex algorithm. MATLAB uses $x = linprog(f, A, b, A_{eq}, b_{eq})$ function and uses dual-simplex by default to find optimized solution.

B. DESIGN OF SECOND STAGE

Now, we introduce the optimization framework to obtain the discrete coefficients η_k for which the passband droop δ is minimized.

1) Estimate the degree of sharpening polynomial N and order of comb Q.
2) A constraint for maximum passband droop δ with equi-ripple in stopband is defined.
3) Create f, A and b using constraints. Then, solve the problem for x. A straight forward way is using MATLAB for linear programming optimization.
4) Obtain the sharpening coefficients η_k.

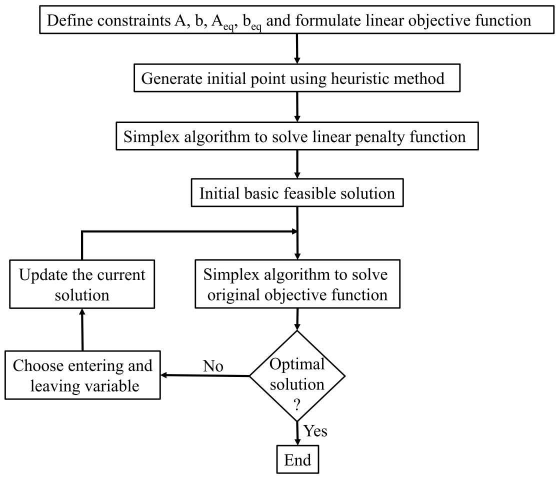


FIGURE 2. **LPO technique based on dual simplex method.**

The sharpening coefficients and appropriate tap locations condition the frequency response of the sharpened structure. We use linear programming optimization in MATLAB to determine these taps and coefficients. Simulations are run to derive a set of coefficients. The proposed second-stage sharpened structure shown in Fig. 3 has transfer function as:

$$H_{22}(z) = \sum_{k=0}^{N} \eta_k z^{-(D_k)} H_2^k(z) \tag{9}$$

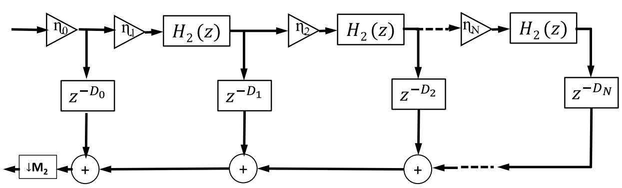


FIGURE 3. **Proposed sharpened second stage H_{22}.**

Here,

$$\begin{aligned} H_2(z) &= \left[\frac{1}{M_2}\frac{1-z^{-M_2}}{1-z^{-1}}\right]^Q; \\ D_k &= \frac{(M_2-1)}{2}(N-k)Q \end{aligned} \tag{10}$$

Given that η_k is any real number, the magnitude response of the above transfer function is given by:

$$\begin{aligned} |H_{22}(e^{j\omega})| &= \sum_{k=0}^{N} \eta_k p^k(\omega), \\ \text{where } \ p(\omega) &= \left|\frac{\sin^Q(\omega M_2/2)}{\sin^Q(\omega/2)}\right| \end{aligned} \tag{11}$$

To calculate the coefficients η_k, we use linear programming, which finds the minimum of a problem specified by:

$$\min_x f^T x \quad \text{such that} \begin{cases} A\cdot & x \le b \\ A_{eq}\cdot & x = b_{eq} \end{cases} \tag{12}$$

Here in our case, we define:

$$\begin{aligned} f^T &= \begin{bmatrix} 1 & 0 & 0 & 0 & 0 & 0 \end{bmatrix} \\ x^T &= \begin{bmatrix} \delta & \eta_0 & \eta_1 & \cdots & \eta_N \end{bmatrix} \end{aligned} \tag{13}$$

where δ determines the passband droop and the cost function f minimizes δ. The equality constraints A_{eq} and b_{eq} are defined for dc frequency as:

$$\begin{aligned} A_{eq} &= \begin{bmatrix} 0 & 1 & 1 & 1 & 1 & 1 \end{bmatrix} \\ b_{eq} &= [1] \end{aligned} \tag{14}$$

A passband droop of δ with equi-ripple in stopband modifies the limits of magnitude response as:

$$\begin{aligned} 1-\delta \le |H_{22}(e^{j\omega})| &\le 1+\delta, \quad \text{where } \ \omega \in \omega_p \\ -\delta \le |H_{22}(e^{j\omega})| &\le \delta, \qquad \text{where } \ \omega \in \omega_s \end{aligned} \tag{15}$$

Based on the above criterion we define constraint matrices as:

$$\begin{aligned} A_1 &= \begin{bmatrix} -1 & p(\omega_1) & \cdots & p^N(\omega_1) \\ -1 & \vdots & \ddots & \vdots \\ -1 & p(\omega_i) & \cdots & p^N(\omega_i) \end{bmatrix} \\ A_2 &= \begin{bmatrix} -1 & -p(\omega_1) & \cdots & -p^N(\omega_1) \\ -1 & \vdots & \ddots & \vdots \\ -1 & -p(\omega_i) & \cdots & -p^N(\omega_i) \end{bmatrix} \\ b_1 &= [1 \quad 1 \quad \cdots]_{i\times 1} \\ b_2 &= [-1 \quad -1 \quad \cdots]_{i\times 1} \\ b_3 &= [0 \quad 0 \quad \cdots]_{2i\times 1} \\ A &= \left((A_1|A_2)_{\omega_i\in\omega_p} \,\middle|\, (A_1|A_2)_{\omega_i\in\omega_s}\right) \\ b &= (b_1|b_2|b_3) \end{aligned} \tag{16}$$

Once we obtain the coefficients after running $x = linprog(f, A, b, A_{eq}, b_{eq})$ function in MATLAB, these are rounded-off to express them as SOPOT. For low complexity we consider $Q = 2$. Simulations are run for different values of N. Fig. 4 shows effect of varying degree (N) and for flat passband we consider $N = 4$.

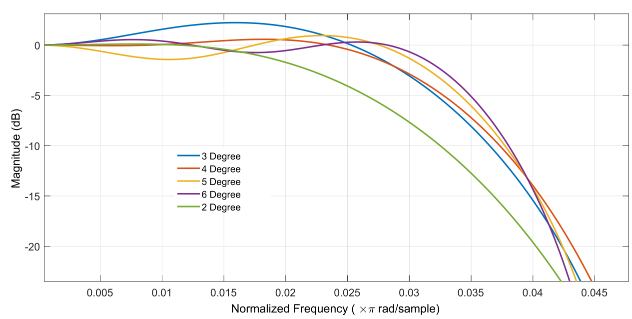


FIGURE 4. **Comparison of magnitude response for various degree(N) for decimation factor $M = 32, M_2 = 8$.**

Table 1 shows sharpening coefficients of second stage.

TABLE 1. Second stage sharpening coefficients of proposed decimation filter(N = 4,Q = 2).

Second Stage coefficients	Value	Number of adders/subtractors
η_0	0	0
η_1	$-1 = -2^0$	1
η_2	$19 = (2^4 + 2^1 + 2^0)$	2
η_3	$-27 = -(2^5 - 2^2 - 2^0)$	3
η_4	$11 = (2^3 + 2^2 + 2^0)$	2

C. COMPUTATIONAL COMPLEXITY

The computational complexity for CIC structure is given by:

$$APOS_{CIC} = Q(M+1) \tag{17}$$

The computational complexity of a sharpening structure in Additions Per Output Sample (APOS) for comb filters is given as

$$APOS_{sh} = NQ(M+1) + P - 1 + \sum_{l=1}^{P} A\left(\eta_{k_l}\right) \tag{18}$$

Here,

N is degree of the sharpening polynomial used in that structure.

Q is order of comb.

P is number of nonzero sharpening coefficients.

A indicates the number of adders required to implement the sharpening coefficients.

The overall computational complexity of proposed structure is given as:

$$\begin{aligned} APOS_{shP} &= APOS_{CIC1} + APOS_{sh2} \\ &= [Q(M_1+1)] \\ &\quad + \left[NQ(M_2+1) + P_2 - 1 + \sum_{l=1}^{P_2} A_2\left(\eta_{k_l}\right)\right] \end{aligned} \tag{19}$$

For example, the proposed filter uses $Q = 2, P_2 = 4$, $N = 4$ for $M = 32, M_1 = 4$ and $M_2 = 8$ and total number of adders to implement sharpening coefficients $A_2 = 8$, we get 93 APOS.

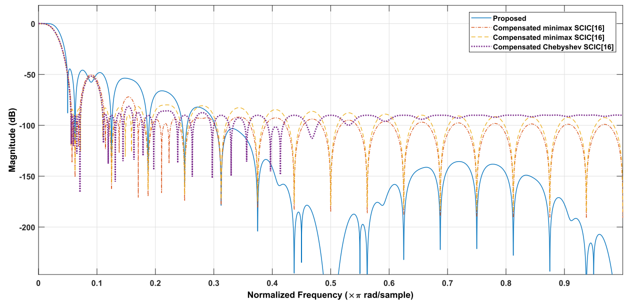
From above equation we can say that applying sharpening technique in second stage results in less complexity as M is divided into smaller parts M_1 and M_2, in comparison to where sharpening is applied directly to comb filter having decimation factor M.

IV. RESULTS AND DISCUSSION

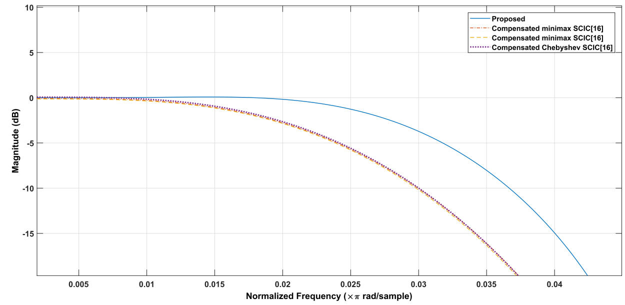
For decimation-by-32, let's say we choose $M_1 = 4, M_2 = 8$ for proposed decimation filter. Fig. 5 and Fig. 6 shows comparison of magnitude response of proposed filter with existing filters.

A. COMPARISON WITH REF [16]

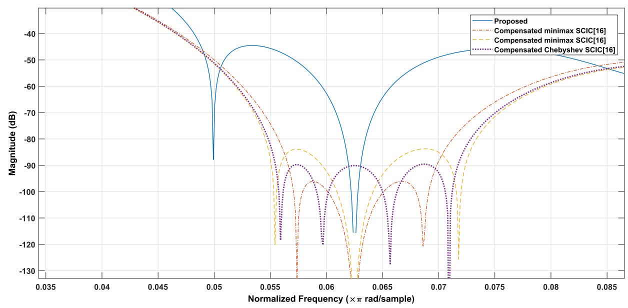
In Ref [16], compensators are designed for sharpened comb filter. The existing two-stage designs with compensators are designed for $M = 32$ only and uses different sharpening polynomial $p(x)$ for different compensators.



(a) Magnitude Response



(b) Passband Magnitude Response (zoomed view)



(c) Stopband Magnitude Response (zoomed view)

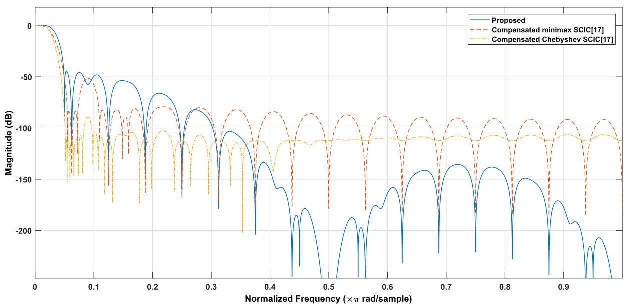
FIGURE 5. Comparison of magnitude response of proposed decimation filter with SCIC filters [16] for decimation-by-32.

Complexity Analysis: One of the sharpening polynomial for compensated minimax SCIC is $p(x) = x^2 - 2^{-7}x$. Thus, degree of sharpening polynomial is 2. Order of comb filter is 2. Number of sharpening coefficients are $P = 2$ and total number of adders to implement sharpening coefficient (1,-2) is $A = 2$. Total number of additional adders for compensator is $N_A = 7$. Overall APOS of Compensated minimax SCIC is given as

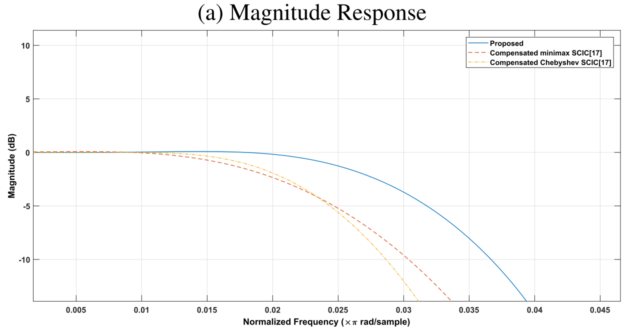
$$APOS = 4(M+1) + P - 1 + \sum_{l=1}^{P} A\left(\eta_{k_l}\right) + N_A \tag{20}$$

B. COMPARISON WITH REF [17]

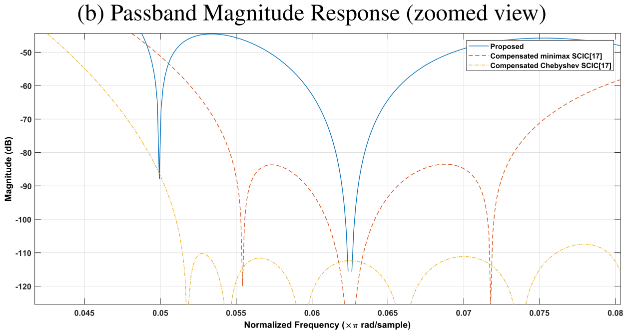
In Ref[17], compensators are designed for sharpened CIC. **Complexity Analysis:** The sharpening polynomial for compensated minimax SCIC in Ref[17] is $= x^4 - 2^{-6}x^2$. Thus, degree of sharpening polynomial is 4. Order of comb filter is 1. Number of sharpening coefficients (P) is 2 and total number of adders to implement sharpening coefficient (1,-2)



(a) Magnitude Response



(b) Passband Magnitude Response (zoomed view)



(c) Stopband Magnitude Response (zoomed view)

FIGURE 6. Comparison of magnitude response of proposed decimation filter with SCIC filters [17] for decimation-by-32.

is $A = 2$. Total number of additional adders for compensator is $N_A = 4$. Overall APOS of Compensated minimax SCIC is given as

$$APOS = 4(M+1) + P - 1 + \sum_{l=1}^{P} A\left(\eta_{k_l}\right) + N_A \quad (21)$$

C. COMPARISON WITH REF[18]

In Ref [18], compensator is designed for comb filter using particle swarm optimization (PSO) technique for passband ripple less than 0.02 dB.

Complexity Analysis: The decimation factor is 32 and the order of filter is 5. Total number of additional adders required for compensator is $N_A = 23$. Overall APOS of compensated CIC is given as

$$APOS = 5(32+1) + N_A \quad (22)$$

Table 2 compares the passband droop and corresponding alias-rejection for the proposed and existing decimators for decimation-by-32. The proposed decimation filter has passband droop $\leq$ 0.007 dB with stopband attenuation $\geq$ 44.91 dB, which is sufficient for many applications like SDR (Software defined radio) receiver [3], [21], [22] and sigma-delta ADC [2]. Other applications of SRC (sample rate

TABLE 2. Comparison of magnitude response parameters of proposed decimation filter with existing filters for decimation-by-32.

Design	Passband Droop (in dB)	Alias Rejection(in dB)
Compensated minimax SCIC [16] $L=3, p(x)=x^2-2^{-7}x, \omega_p=\frac{\pi}{5}$	0.04	86.0
Proposed	0.0014	48.37
Compensated minimax SCIC [16] $L=3, p(x)=x^2-2^{-6}x, \omega_p=\frac{\pi}{4}$	0.09	82.5
Proposed	0.007	44.91
Compensated Chebyshev SCIC [16] $L=3, p(x)=2^{15}x^2-2^9x+1$, $\omega_p=0.226\pi$	0.06	90.2
Proposed	0.002	46.42
Compensated minimax SCIC [17] $L=3, p(x)=x^4-2^{-6}x^2, \omega_p=\frac{\pi}{4}$	0.03	74.34
Proposed	0.007	44.91
Compensated Chebyshev SCIC [17] $L=3, p(x)=2^{15}x^4-2^9x^2+1$, $\omega_p=0.226\pi$	0.02	38.12
Proposed	0.002	46.42
Compensated CIC based on PSO [18]	0.02	–
Proposed	0.007	44.91

TABLE 3. Complexity analysis in terms of APOS.

Design	APOS	APOS for M=32	Complexity Reduction
Proposed	$Q(M_1+1)+LK(M_2+1)+P-1+\sum_{l=1}^{P}A(\eta_{k_l})$	93	–
Compensated minimax SCIC [16] $p(x)=x^2-2^{-7}x$	$4(M+1)+P-1+\sum_{l=1}^{P}A\left(\eta_{k_l}\right)+N_A$	142	34.5%
Compensated Chebyshev SCIC [16] $p(x)=2^{15}x^2-2^9x+1$	$4(M+1)+P-1+\sum_{l=1}^{P}A\left(\eta_{k_l}\right)+N_A$	144	35.4%
Compensated minimax SCIC [17] $p(x)=x^4-2^{-6}x^2$	$4(M+1)+P-1+\sum_{l=1}^{P}A\left(\eta_{k_l}\right)+N_A$	139	33%
Compensated Chebyshev SCIC [17] $p(x)=2^{15}x^4-2^9x^2+1$	$4(M+1)+P-1+\sum_{l=1}^{P}A\left(\eta_{k_l}\right)+N_A$	138	32.6%
Compensated CIC based on PSO [18]	$Q(M+1)+N_A$	188	50.5%

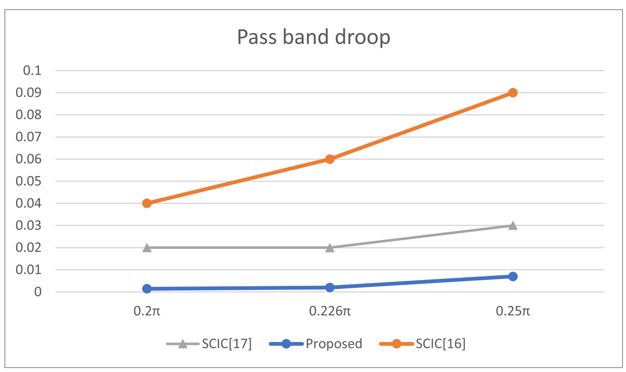


FIGURE 7. Comparison of passband droop.

conversion) filter includes image filtering [23], digital audio resampling [24] and continuous-time signal processing [25].

Table 3 compares the computational complexity in terms of APOS for decimation by 32. The compensated CIC based on PSO in Ref [18] has highest complexity. As seen the proposed design has least passband droop with suitable alias rejection and low computational complexity. Proposed decimation filter shows 65% improvement in passband droop with 50.5% less complexity when compared with compensated CIC based on PSO. Fig. 7 shows comparison of passband droop for various cut off frequencies and proposed filter shows minimum passband droop.

V. CONCLUSION

This proposed work considers the optimal design of decimation filter and compared its performance with other approaches like particle swarm optimization (PSO) based CIC compensator and compensated SCIC. Filter coefficients have been obtained by linear programming optimization (LPO) technique. It is observed that proposed filter provides a better solution in terms of computational complexity, passband droop and eliminates the need of compensator. This filter can be employed in applications that demand lesser passband ripples and stopband attenuation.

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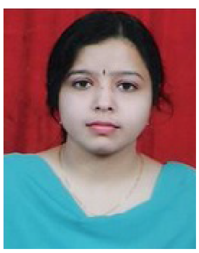
DIVYA GAUTAM received the B.Tech. degree in electronics and communication engineering in 2010, and the M.Tech. degree in VLSI Design from CDAC, Noida, in 2013. She is currently pursuing the Ph.D. degree with the Department of Electronics and Communication, Maulana Azad National Institute of Technology (MANIT), Bhopal, India. She has two years of teaching experience. Her areas of interest are digital signal processing, digital communication, and VLSI design.

KAVITA KHARE received the B.Tech. degree in electronics and communication engineering in 1989, the M.Tech. degree in digital communication systems in 1993, and the Ph.D. degree in VLSI design in 2005. She is a Professor with the Department of Electronics and Communication Engineering, Maulana Azad National Institute of Technology (MANIT), Bhopal, India. She has nearly 100 publications in various international conferences and journals, and the publications topics, include CMOS logic circuits, logic gates, leakage currents, low-power electronics, MOSFET circuits, VLSI, digital arithmetic, digital computers, direct digital synthesis, integrated circuit design, logic design, modulators. Her fields of research interest are VLSI design, communication systems, and design of arithmetic circuits and various communication algorithms related to synchronization, estimation, and routing. She is a Fellow of IETE, India, and a Life Member of ISTE.



BHAVANA P. SHRIVASTAVA (Fellow, IEEE) received the bachelor's degree in electronics and communication engineering in 2003, the M.Tech. degree in digital communication systems in 2007, and the Ph.D. degree in VLSI design in 2014. She is an Assistant Professor with the Department of Electronics and Communication Engineering, Maulana Azad National Institute of Technology (MANIT), Bhopal, India. Her fields of research interest are VLSI design and communication systems and networking.