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FIR filter-based Sample Rate Convertors and its use in NR PRACH

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Abstract - *The transmission of data through the air interface using both Physical Random Access channels (PRACH) and Physical Uplink Shared channels (PUSCH) in an Orthogonal Frequency Division Multiplexing (OFDM) system The data rate at instances of PRACH subcarrier frequencies is smaller than that of PUSCH subcarriers. To achieve seamless data transmission, sampling rate conversion is employed to match the varying subcarrier spacings and channel bandwidths. Sampling rate conversion involves adding or removing samples while preserving the duration and shape of the signal. This process introduces variations in the power spectral density in the frequency domain, necessitating filtering techniques to retain the original power spectral density. Specifically, in NR PRACH processing, the IFFT size at the transmitter and FFT size at the receiver are fixed based on the PRACH sequence length, requiring sampling rate conversion to align with the IFFT and FFT processing. This paper discusses and compares the characteristics and performance results of the digital Finite Impulse Response (FIR)-based sampling rate conversion filter, the Cascaded Integrator Comb (CIC) Filter, which is used with interpolation and decimation filters at the transmitter and receiver, respectively. The study provides insights into selecting the most suitable sampling rate conversion filter for OFDM systems, ensuring efficient data transmission, and optimizing spectral efficiency.*

Key Words: OFDM, PRACH, PUSCH, sampling rate conversion, interpolation, decimation, FIR filter, CIC Filter, spectral efficiency, wireless communication

1.INTRODUCTION

Sampling rate conversion is a fundamental process in signal processing that allows for the addition or removal of samples while preserving the signal's duration and shape. Its significance spans various industries, including wireless communications, where precise sampling rate conversion plays a vital role. An example is in Orthogonal Frequency Division Multiplexing (OFDM) systems, where accurate conversion ensures compatibility between the fixed Inverse Fast Fourier Transform (IFFT) at the transmitter and the Fast Fourier Transform (FFT) at the receiver. This compatibility is critical for efficient New Radio (NR) Physical Random-Access Channel (PRACH) processing, where the IFFT and FFT sizes are determined based on the PRACH sequence length.

In this research paper, the Cascaded Integrator Comb (CIC) filter, recognized as a digital Finite Impulse Response (FIR) filter, is thoroughly examined for its ability to achieve precise and distortion-free sampling rate conversion in communication systems based on Orthogonal Frequency Division Multiplexing (OFDM). The CIC filter is highly favored for such applications due to its straightforward operation and minimal computational complexity.

To align the sampling rate with the set IFFT FFT processing requirements in the context of OFDM systems, sampling rate conversion is required. The sampling rate is determined by subcarrier spacing and channel bandwidth; any deviation from these parameters necessitates conversion of the sampling rate to OFDM. The IFFT size at the transmitter and FFT size at the receiver are predefined based on the length of the PRACH sequence, which further emphasizes the significance of sampling rate conversion in NR PRACH processing.

The proposed technique converts sampling rates using decimation filters at the receiver and interpolation filters at the transmitter. These filters effectively increase or decrease the number of samples to achieve the desired sampling rate while preserving the original signal's duration and shape. The primary goal of this paper is to evaluate the CIC Filter's capabilities and characteristics in the context of sampling rate conversion in OFDM systems.

The paper's structure is as follows: Section 2 provides a comprehensive literature review of sampling rate conversion methods and filters utilized in OFDM systems. In Section 3, the methodology is presented, which includes an overview of digital FIR-based filters and the specific requirements for sampling rate conversion in OFDM systems and section 4 is having results. Finally, Section 5 concludes the paper by summarizing the key findings and proposing potential avenues for future research.

2. LITERATURE SURVEY

In his article titled "CIC Filter Introduction," Matthew P. Donadio [3] highlights the increasing importance of cascaded integrator-comb filters in modern Digital Signal Processing (DSP) systems. As data converters continue to advance in speed, the need for narrow-band extraction from wideband sources and narrow-band construction of wideband signals becomes critical. Accomplishing these tasks relies on fundamental signal processing techniques like decimation and interpolation. Despite the advancement in digital hardware speed, the demand for efficient solutions persists. It was Hogenauer [1] who developed the CIC filters to address this challenge. These versatile and multiplier-free filters are well-suited for hardware implementation and can handle arbitrary and significant rate changes effectively.

The CIC filter can be efficiently implemented using a moving average filter with NR taps, followed by output decimation by a factor of R. Similarly, an interpolating CIC filter utilizes an NR tap moving average filter operating at the output sample rate, with R-1 zero samples inserted between each input sample [2]. The beauty of CIC filters lies in their simplicity, as they exclusively employ additions and subtractions for their arithmetic operations. As described by the authors, CIC filters are technically lean, mean, and efficient filtering machines, making them highly effective in their functionality.

Researchers have optimized the hardware required for implementing CIC decimator and interpolator structures. Through a combination of modal analysis, MATLAB simulations, and FPGA synthesis using Xilinx ISE, they validate their results while exploring different rate factors and stage configurations. The device utilization summary for decimators and interpolators with varying stages is presented using the Xilinx Spartan 3 FPGA [4].

The challenge of handling high-frequency transmission signals and implementing decimation with FIR or IIR filters arises due to the significant cost associated with purchasing numerous multipliers. In response to this challenge, a practical solution has been devised in the form of a five-stage CIC filter that eliminates the need for multipliers. This innovative structure finds its primary application in Digital Signal Processing (DSP) applications [5]. By using the CIC filter without multipliers, the cost and complexity of the implementation are significantly reduced while maintaining effective decimation of high-frequency signals.

The efficiency of conversions by odd factors compared to even factors in the class of linear-phase FIR Mth-band filters has been demonstrated. For odd factors, the proposed structures are more efficient and can be derived from conventional two-stage structures. In contrast, recently created add-equalize structures, which are also capable of arbitrary-integer conversion, exhibit comparable complexity to conventional multi-stage converters (when applicable), but are not more effective for odd M compared to even M [6]. The findings indicate that the proposed structures outperform these add-equalize structures, especially for odd conversion factors, offering improved efficiency in the context of linear-phase FIR Mth-band filters.

Jun Woo Kim et al. [7] introduced a reliable PRACH detection method that remains dependable regardless of the search time offset. While this method takes twice as long to search for PRACH, the latency of PRACH detection is not a significant concern in modem design since the detection process occurs only once per frame. Despite the longer search time, the PRACH detector proposed in the study accurately identifies the optimal UL (Uplink) start point for the anticipated channel model in the intended service, as confirmed by simulation results. This highlights the effectiveness and precision of the proposed PRACH detection method.

Tuan Anh Pham and Bang Thanh Le present an enhanced algorithm in their work [8], aimed at effectively mitigating false alarm peaks caused by various factors, such as noise and adverse channel conditions like multipath fading, Doppler shift, frequency offset, and timing offset. The proposed method employs multiple detection steps to achieve this objective. By optimizing detection thresholds, the algorithm fulfills the system's specific requirements concerning detection probability and false alarm probability, especially under different Signal-to-Noise Ratio (SNR) levels and varying propagation conditions. Through this approach, the authors successfully address the challenges posed by noisy and dynamic channel environments, enhancing the overall performance of the system.

A continuous-time signal is a signal that is present at all points in time because there are an infinite number of time instances separating any two points in space. A continuous-time signal is sampled at specific, predetermined time instants to produce a discrete-time signal, in contrast. The corresponding samples represent the values of the signal, and the sample times indicate when the signal is defined.
The Nyquist-Shannon sampling theorem mathematically limits the sampling rate by stating that the minimum sampling rate should be at least twice the maximum frequency of the signal being sampled. As a result, it is possible to successfully recover the original continuous-time signal.

Consider, for instance, a signal that continuously changes in frequency from f1 to f2, with f1 being the lowest frequency and f2 being the highest. The minimum sampling frequency needed to accurately recover the original continuous-time signal is 2*f2.

With the aid of various signal processing techniques, such as sample and hold circuits and low-pass filters, the original continuous-time signal can be recovered from its samples.

The particulars of these methods are outside the purview of this discussion.

Handling high-rate samples requires more hardware memory and processing power. Modifying the sample rate becomes a crucial technique in signal processing systems, depending on the hardware resources available. When operating with constrained hardware, lower sample rates are preferred. This method is useful in fields like image scaling and audio/visual systems, where various sampling rates may be used for technical, practical, or even historical reasons.

The process of changing a discrete signal's sampling rate or frequency to produce a new discrete representation of the underlying continuous signal is known as sample-rate conversion. Interpolation, or upsampling, refers to the process of going from a lower rate to a higher rate by adding more samples to guarantee better signal quality. On the other hand, decimation or downsampling refers to the removal of samples when going from a higher rate to a lower rate to save memory, processing power, and time resources.

2. METHODOLOGY

2.1 Sampling rate conversion's effects on signal's time-frequency characteristics

Sampling rate conversion has a significant impact on a signal's time and frequency domains. To be successful, the original signal's desired properties must always be maintained.

Interpolation:
Time domain impacts:

When interpolation is done, extra samples are added to the original sample set, increasing the information content and possibly raising the signal's quality. However, because there are more samples, this also results in more storage space being needed and slower processing.

Frequency domain impacts:

The frequency-domain characteristics of the signal can be significantly altered by interpolation as well as decimation. As more samples are added, the interpolation process introduces amplitude variations in the newly inserted samples. At integer multiples of the original sampling rate, copies of the original spectrum start to appear as a result.

Zero-valued samples are first added between the original samples to begin the interpolation process. The original spectrum is reproduced at integer multiples of the original sampling rate as a result of these abrupt amplitude variations. This phenomenon can be seen in the upsampled signal's spectral features, where the original spectrum's copies at these integer multiples become very obvious.

Decimation:
Time domain impacts:

Decimation involves removing samples from the original sample set, which could lead to a decrease in signal quality and information content. However, it has benefits like less storage space and quicker processing.

Frequency domain impacts:

Decimation also affects the signal's spectral characteristics. Making sure that the newly reduced sampling rate can faithfully capture all of the spectral properties of the original signal presents a challenge during decimation. Aliasing can happen if the original signal contains frequency components that are higher than the range that can be represented by the slower sampling rate. Aliasing is the process of folding frequency components back into the relevant band, which can lead to distortions in the signal's spectral properties.

For example, when downsampling a signal by a factor of four, if the original signal contains a frequency component (CIC) filter, The CIC filter can deal with indeterminate and large rate changes effectively.

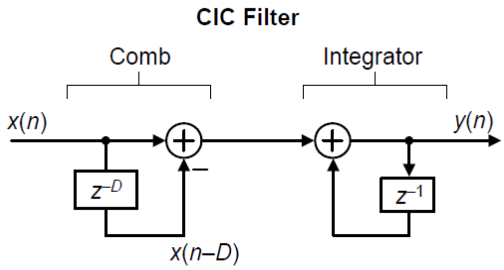

Fig -1: A Cascaded integrator-comb implementation of a D-point recursive running sum filter.

The comb section and the integrator are the two fundamental building blocks that make up the CIC filter. The feedforward portion, also referred to as the comb section, includes a differential delay denoted by the letter D. The integrator, on the other hand, refers to the feedback section. The current input sample is subtracted from a delayed input sample in the comb stage. The integrator functions as an accumulator.

The difference equation representing the time-domain behavior of the CIC filter is given by:

$$y(n) = x(n) - x(n-D) + y(n-1) \quad (1)$$

Mathematically, the integrator and comb circuits can be as follows, where x [n] is the input and y [n] is the output, with a differential delay of D:

Integrator: $y[n] = y[n-1] + x[n]$ (2)

Comb: $y[n] = x[n] - x[n-D]$ (3)

2.2 CIC filter for Interpolation and Decimation

The CIC filter provides a flexible solution for rate conversion in DSP systems when significant rate changes are necessary. Due to its hardware compatibility and multiplier-free implementation, it is a desirable choice for efficient and effective signal processing.

CIC filters serve two main purposes: anti-imaging filtering for interpolated signals and anti-aliasing filtering for decimated signals. In the decimation process, denoted as R, the sample rate fs, out becomes fs, in/R after keeping only every Rth sample. When used for interpolation, the CIC filter performs upsampling by inserting R-1 zeros between each x(n) sample, resulting in an output sample rate of fs, out = Rfs, in. This interpolation involves inserting zeros, often termed "zero stuffing," followed by low-pass filtering using a CIC filter.

The figure below illustrates a single-stage CIC filter capable of performing interpolation or decimation. Notably, the comb section on the lower sample rate side of the filter has a reduced delay length (differential delay) of N = D/R. This is because a multi-stage comb delay before R downsampling is equivalent to an N-sample comb delay after R downsampling. Similarly, an N-sample comb delay before upsampling by R is equivalent to a D-sample comb delay after upsampling by R for the interpolation filter. This configuration enables efficient and accurate CIC filtering for both interpolation and decimation operations.

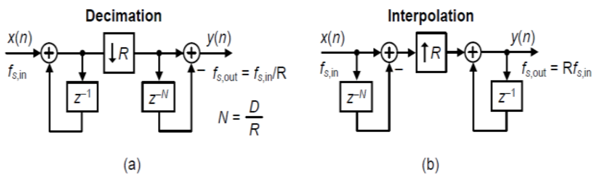

Fig 2: A single-stage CIC filter for decimation and an interpolation filter.

The most popular method is to cascade multiple CIC filters to improve the anti-aliasing and image-reject attenuation of CIC filters. The combined transfer function in the z-domain of an Mth-order CIC filter with M stages of comb and integrator is obtained by multiplying their transfer functions.

$$H_{\text{CIC, Mth-order}}(Z) = \frac{W(z)}{X(z)} = \left|\frac{\sin(2\pi f D/2)}{\sin(2\pi f/2)}\right|^{M} \quad (4)$$

When using the x(n) input to the w(n) sequence, the filter's frequency magnitude response is as follows:

$$|H_{\text{CIC, Mth-order}}(e^{j2\pi f})| = \left|\frac{\sin(2\pi f D/2)}{\sin(2\pi f/2)}\right|^{M} \quad (5)$$

2.3. CIC interpolator comparison in NR PRACH

To handle the change in sampling rate between the PRACH and UL (Uplink) data channels, interpolation is required in the 5G NR PRACH (Physical Random Access Channel) transmission chain. While it would be ideal, due to complexity and memory constraints, to have an IFFT or FFT size that corresponds to the number of IQ (In-phase and Quadrature) samples to be processed, It is therefore necessary to use an interpolator or decimator.

For instance, the interpolation factor is determined using the data channel's subcarrier spacing of 15 kHz in the case of long PRACH Format 0, where the subcarrier spacing is defined as 1.25 kHz. For PRACH, the IFFT size and the number of IQ samples can change depending on the bandwidth or sampling rate, and this can affect the interpolation factor. Given that PRACH transmission and reception occur at the same sampling rate, the reciprocal of the interpolation factor must be used as the decimation factor at the PRACH receiver.

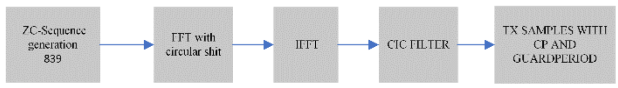

Fig 3: Block diagram of PRACH flow with CIC filter

The NR PRACH TX (transmit) flow is shown in the flow diagram, along with the interpolation filters that are used. CIC (Cascaded Integrator Comb) interpolators have advantages over other interpolation methods, such as polyphase filtering, in terms of complexity and memory requirements. A CIC filter has a fixed number of stages and taps, typically five stages, which enables efficient implementation without the need to store filter coefficients for various scenarios. Furthermore, CIC filters do not require the complex multiplications necessary for polyphase filtering; instead, they only use fixed-point addition and subtraction. CIC filters are therefore highly suggested for DSP (Digital Signal Processing) systems because they provide computational efficiency and equivalent performance to polyphase filtering with lower complexity and memory requirements.

3. RESULTS

Table -1: Simulation Parameters

Parameters	Value
PRACH Format 0	1.25KHz SCS
Bandwidth	20MHz
Sampling rate	30.720ksps

IFFT size	2048
No. of IQ samples after Up-sampling	24576 (as per 3GPP 38.211)
No.of CP samples	3168 (as per 3GPP 38.211)
Interpolation factor (R)	12

In figure the frequency response of CIC filter is sharp and the phase is liner compared to polyphase filter. By this we can say that the CIC filter has the better frequency response than polyphase filter.

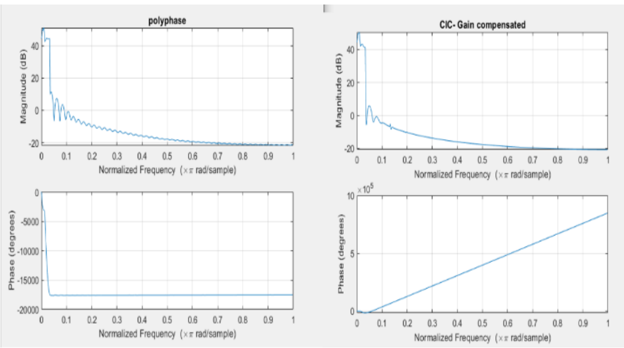

Fig 4: Frequency response plot comparison between Polyphase and CIC

3. CONCLUSIONS

By incorporating the CIC Up-sampling filter in PRACH, the problem of frequency mismatch between PRACH and PUSCH subcarriers is resolved. The up-sampling process enables synchronization, aligns the frequencies, and maintains the integrity of the PRACH signal in the time domain. This ensures reliable transmission of data through the air interface using both PRACH and PUSCH channels at the same data rate.

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