



A Review on Design and Applications of Nyquist Filter

Kavita Rani

PG Scholar, Electronics and Communication, Chitkara University, Rajpura, India

Email: kavita.rani@chitkara.edu.in

Deepak Saluja

Assistant Professor, Electronics and Communication, Chitkara University, Rajpura, India

Email: deepak.saluja@chitkara.edu.in

Abstract: A humongous development in the study of Nyquist filter has been observed in the recent years. This paper presents a review of the work done on a Nyquist filter in the last decade. Nyquist filter is called M^{th} band filter because it has zero coefficient at every M^{th} sample except its middle coefficient. Different filter methods have been proposed with the focus of improving filter parameters like gain, efficiency, intersymbol interference, minimized power dissipation. Designing as well as use of Nyquist filter has been reviewed.

Keyword: Please Efficiency; Filter parameters; Intersymbol Interference; Nyquist filter; Timing Jitter;

1. INTRODUCTION

Filtering is the main process in any system. Here in this paper, I have taken papers on Nyquist filter from 1982 to 2016. I have seen that Filters have an important role in all sampled data systems. Most analog-to-digital converters are preceded by a filter which removes frequency components that are outside the ADC's range. I have studied different techniques of designing the Nyquist filter. Every Author has designed their own technique or method to improve the performance of Nyquist filter. The filter is named after the Swedish-US engineer Harry Nyquist (1889-1976). This filter has zero value at every M^{th} ample except its middle value. There are some aspects which must be taken while designing any filter that are power dissipation should be minimized, inter-symbol interference, gain, attenuation and many other factors that we will see in the different research work.

We will see also many applications of Nyquist filter. For any digital model system, digital Nyquist filter plays an important role. There are two methods for designing Nyquist filter that are finite impulse response and infinite impulse response. Where finite impulse response requires Z-transform and infinite impulse response (IIR) filter requires both frequency and time response. because of all pass sections used for timely response, order of the filter increases because of that the filter become more susceptible, which requires a high coefficient for finite impulse response (FIR).

So, to overcome these problems, they have designed an infinite impulse response filter which has zero ISI and small roll off factor and order of the filter can be reduced. for smaller roll off factor for the infi-

nite impulse response (IIR) filter as compared to finite impulse response (FIR) filter. They have given some conditions for zero intersymbol interference and compare the result of finite impulse response (FIR) and infinite impulse response (IIR) Nyquist filter. Nyquist limit is the only limit that can be handled by the maximum frequency component a sampled data system [1]. There is a serious problem in the case of filtering is inter-symbol interference. Inter-symbol interference is a type of failure or disturbance of the signal in which one symbol interferes with symbols presented in the system. When this interference occurs, this leads to make whole system unreliable because the signal or symbol that get interferes with the other, already have noise contents, so it will make whole signal undesirable. If the pulse is spreading beyond the time interval that is allotted to it, it will cause it to interfere with adjacent or other pulses.

In [1], "Nakayam and Mizukami", introduced a method to design infinite impulse response (IIR) filter with zero ISI and reduced hardware size. but that method require arithmetic operation. In [2], Finite impulse response (FIR) Nyquist filter is designed by linear phase. In this method, a new technique is used to design finite impulse response (FIR) filter by cascading finite impulse response (FIR) subfilters. By doing this, they have decreased no. of multipliers. Here each subfilter is also a Nyquist filter. It follows the criterion that is every M^{th} impulse coefficient is zero but middle coefficient is not zero. In [2], they have designed such a filter whose single stage is even more efficient than the filter [1]. If the ISI is detected in the system, it will introduce errors in the decision device at the receiver output. So, the designing of fil-

ter is based on the fact that it will ensure minimum effects of ISI but during this year, this [2] does not provide other features of Nyquist filter such as gain, robustness to timing jitter and efficiency etc.

There are many issues generated while designing Nyquist filter like zero intersymbol interference we have already seen in [1] - [2], but there is another issue that is infinite stop band attenuation. In [3], they have described the conditions for different type of filters such as an infinite impulse response (IIR) Nyquist filter, Finite impulse response (FIR) and raised cosine RC filter. Every filter has their different requirement for their working. They have described that for RC filter, at every point the value of the Impulse response and transfer function is known. But in finite impulse response (FIR) filter, it only can perform either in the time domain or in the frequency domain which depends upon the time samples. In case of infinite impulse response (IIR) filter design, it is not simple as a finite impulse response filter; here it is based upon the least square approximation of time domain where the phase response of the filter must be ignored.

Some problems while designing Nyquist filter are still there that is minimum power dissipation and complexity. In [1] - [3], they were only talking about zero ISI but what about the complexity of filter design, which is the major factor. So, in [4], Ping – Kuen (Andy) Lam, Earl W. McCune and Michael A. Soderstrand designed a method to reduce the complexity which is essential to get out the filter’s impulse response. As we know, transfer function of the filter is even symmetric when frequency (f_0) is equal to zero and it is odd symmetric about cut off frequency $f=f_0$. such a filter is called Nyquist filter. If the sampling frequency receives, $f_s=2f_0$, that type of filter is called Nyquist filter. In [4], Nyquist filter have been designed with several techniques like convolution of two functions, where one of them is brickwall low-pass filter and other is even symmetric and second technique is using window function.

In 1998[5],” B. Farhang-Boroujeny and George Mathew “Proposed a paper a technique in which researcher has designed a Nyquist filter with robustness performance against Timing Jitter. As we know, the main problem in data communication is the design of pulse shapes that results in zero intersymbol interference (ISI) while having good robustness to timing jitter and sufficient stopband attenuation. From Fourier transform $H(f)$ of the overall transmitter-receiver filter response, an important and sufficient condition for zero intersymbol interference is, Nyquist criterion given as

$$\sum_{m=-\infty}^{\infty} H(f + (m / T)) = T,$$

Where T is the symbol period. The time-domain equivalent of this criterion is

$$h(kT) = 1, \text{ if } k = 0 \\ = 0, \text{ otherwise.}$$

Where $h(t)$ is the continuous-time impulse response of $H(f)$. They have taken some aspects to design this type of filter some of them are length of the filter is finite for physical realizability, they have considered zero impulse response at nonzero integer multiples of the symbol period (T) for getting zero ISI. So, to overcome from the problems, they proposed a solution using the minimax design approach in conjunction with a novel windowing technique.

There are many applications of Nyquist filter and every filter has their different requirements. In [6], the design of Nyquist filter is introduced with zero intersymbol interference and low group delay where group delay is half the filter order for linear phase FIR. In some real time signal processing system, low group delay is required. There are two disadvantages of Nyquist filter:

- a) Stability
- b) More time consumption

In [6], a methodology is implemented to overcome these problems.

2. NYQUIST FILTER FROM 2004 TO 2010

One another factor which plays vital role in better performance of any filter is power dissipation so to minimize the power dissipation of analog channel select filter “Jarkko Jussila, Kari Halonen” Introduce a method in [7] for minimization of power dissipation of analog channel select Filter and Nyquist rate A/D convertor in ultra/Fdd .In this paper some equations are derived that tell us about the relations between the order of the filter, the sample rate and resolution of the ADC, and the total power consumption .

$$n_p(f_s) \geq \left\lceil \frac{3.3}{\log_{10}\left(\frac{f_s}{2MHz}\right)} \right\rceil \quad (1)$$

$$L(f) = n_p \cdot \log_{10}\left(\frac{f}{2MHz}\right) \cdot 20db \quad (2)$$

$$N = N_{S+N} + \left\lceil \frac{\log_{10}\left(1+10^{(5.25 db - L(10MHz))/20}\right)}{\log_{10} 2} \right\rceil \quad (3)$$

$$P_{D,TOT}(n_p, f_s) = P_D \cdot POLE \cdot n_p(f_s) + E_{conv} \cdot 2^{N(n_p(f_s))}. \quad (4)$$

Where N is the number of bits, f_s is the sample rate, and E_{conv} is the energy required for a single conversion. The best reported value of E_{conv} is 0.5pJ and the average is approximately 5pJ. But all aspects for designing of Nyquist filter was not properly defined in this paper. The Equations (1) to (4) are derived to minimize the power dissipation [7]. These equations are used to approximate the value of n_s and f_s which provide the lowest power. There are many problems generated while designing the Nyquist filter which has been reported in [1]-[7], Main challenges in designing of Nyquist filter is basically to find out the finite set of coefficient which satisfies intersymbol interference, low sensitivity to timing jitter and good stop-band elimination.

In [5], “B. Farhang-Boroujeny and George Mathew” has designed a technique for Nyquist filter which compromises the above criteria but when a filter is designed with finite word length then that consumes more time and become very complicated. So, to reduce this type of complexity, a new method was introduced in [8].

In 2007 [9], The Author has proposed a Square-Root Nyquist M^{th} Filter Design for Digital Communication Systems. In [9], filter have been designed such that it should meet the condition of Nyquist criterion such as design a filter $H(z)$. While designing the filter response, the various aspects that may be considered

- a) When channel distortion is not present, so to minimize the interference among data symbols, one condition that should be minimized, is Nyquist criterion.
- b) There are two system parameters which are dictated by frequency mask in appropriate standards. So that the filter that was designed must fit within that mask.
- c) The side-lobe's magnitude of the impulse response should be reduced, to solve the difficulty of timing jitter.
- d) To minimize implementation cost filter's length should be small and filter should be linear phase filter.
- e) Tail size of the pulse shape should be reduced to get the modulated signal with reduced peak-to-average power ratio. These types of modulated signals have many applications like these signals are used in power amplifiers.

The technique of designing Nyquist filter which is offered in this paper does not take the impact of analog filters. Because since digital domain is considered as the main field. Analog filters are only applied at an intermediate frequency, or we can say that at the ra-

dio frequency (RF) band, but the baseband signals have already interpolated with high value of interpolation factor. In such systems, analog filters may be ignored because these filters have no major impact on the equivalent baseband channel model while designing the pulse shaping filters. By using the techniques offered in this novel, we can design a filter with superior performance and lower computational complexity. This algorithm has ability to design the filters that are of linear phase. Through this proposed method Nyquist filter can be designed with robustness to timing jitter and peak to average ratio can be reduced [6].

In 2009 [10], “Nicolae Dumitru Alexandru, Felix Diaconu” investigates a new type of Nyquist filters to improve intersymbol interference, its performance depends upon signal to noise ratio of the channel, in this method a new technique is used that is the frequency characteristics are derived from an ideal staircase frequency characteristic with twelve levels. These frequency characteristics are derived by using interpolation with linear function which passes through interpolation points.

3. NYQUIST FILTER FROM 2010 TO 2014

In 2011[11], “Alexandra Ligia BALAN, Nicolae Dumitru Alexandru” has proposed a technique to design Nyquist filter with some improvement. Improved Nyquist filter means a filter with minimum distortion and less probability error in the presence of timing jitter. The method introduced [11] in which frequency characteristics are represented by staircase. Those characteristics are only for positive frequencies. In those characteristics, interpolation has done with polynomial function.

In [11], some conditions are introduced to reduce errors and the author has also described some parameters for designing of Nyquist filter such as height & width of the rectangular pieces. 'k' is the index of these parameters, on which shape of the characteristics depends. In the same year [12], N.D. Alexandru and A.L. Balan has done improvement in [11]. The author has designed improved Nyquist filter by using piecewise linear characteristics with the segment of changeable slopes and length in frequency domain is presented. The author has investigated new parametric pulses for specific values of design parameters which represents slighter fault in the existence of symbol timing error.

In 2012 [13], for all the problems such as intersymbol interference, signal to noise ratio, timing jitter, error probability, the author has designed a solution. That solution is designed Nyquist pulse which is based on a piecewise rectangular In which the frequency characteristics are of concave shape of the transition region. In [13], the new characteristics are defined for positive frequencies. In this design, two rectangular and two sinusoidal pieces are chosen. As

compared to [12], this design requires less number of parameters and mathematical expressions of frequency and time characteristics have less complexity. In [1]-[12], we have seen that filters are basically affected by multipath propagation, tracking error, Doppler effect, the resultant of these problems is known as 'jitter', due to which amplitude of the signal get reduced. So, in 2013, Author has proposed a method to construct a filter in which its overall frequency is split between transmitter and receiver and also in this method, an additional delay has been introduced for physical realization. Some equations are also defined for frequency characteristics. Here probability error can be easily, accurately and quickly calculated in [14]. In [11] -[12], Nyquist filter has been designed with different methods like piecewise linear interpolation or interpolation with polynomial function. But when these techniques are sampled with time offset, the performance of the filter is not too good. Whereas in [15], performance of the filter when sampled with time offset, is better than [11] - [12].

4. NYQUIST FILTER 2015 ONWARDS

In 2015 [16], "Weili Yan, Chunling Du, and Chee Khiang Pang" has designed a type of Nyquist filter which is in feedforward form to attenuate the disturbance [16]. This filter is driven by LMS algorithm, through which all effects of input disturbance has been cancelled out. This proposed method can successfully overthrow the disturbance to Nyquist filter and outside that.

In [6], "Xi Zhang and Toshinori Yoshikawa" have proposed a technique for low group delay, but the author did not explain anything regarding high stop band attenuation. To bound the quantity of out of band spectrum leakage, there is a requirement of high stop band attenuation in the case of designing square root Nyquist filter. As in [1] -[10], different methods have been proposed for zero intersymbol interference, timing jitter, power dissipation. but that methods are limited to roll-off factor, $\alpha = 1$. So, for higher value of rolloff factor in [17].

The author has proposed a designed for small group delay and high stop band reduction for the square root Nyquist filter. Here in [17], inverse fourier transform technique is used for the finite impulse response which is generated from a frequency mask. There are three conditions are given for higher values of roll-off factor for getting low group delay and high stop band attenuation. One of these conditions is frequency domain design procedure in which frequency mask H_1 is taken over $N_{DFT} = 2(L_1 + L_2 + A + 1)$ Points where $-\frac{N_{DFT}}{2} \leq i \leq \frac{N_{DFT}}{2}$, where A and L_1 are the parameters on which roll off factor depends.

The author has also given the equation for roll off factor in terms of A and L_1 . In [17], an example is taken off frequency mask by keeping $\alpha = 0.25$ and

oversampling rate is 2 and analyze the result. Other condition or method is a coefficient computation procedure, which gives high and fast decaying stop band attenuation.

5. IMPLEMENTATION OF NYQUIST FILTER

In [1]-[17], Authors have designed the Nyquist filter with their own techniques. Some applications are also available in which this filter is used for different reasons. Such as use of Nyquist filters in Image processing, Speech processing, etc. Filters are also designed for speaker identification [18]-[20]. Every year some improvement has done with this filter for speaker recognition. While designing Nyquist filter for such application Author has ensured that the precision should be highest, but computational efforts should be fewer. The author has designed Mel-frequency Cepstral coefficients, in which Nquist filter is used for speaker identification [18].

This technique is used for low frequencies, here low frequency contains speaker specification information, but for high frequency this scheme cannot be taken into consideration. So, to get higher accuracy in case of text independent speaker identification, in [19] the Author has designed improved Nyquist filter. This method is used for low and high frequency both. Here these frequencies consist of speaker specific information. This method gives high resolution at low and high frequency zone. For the same application of Nyquist filter, some improvement has done in [20]. Nyquist filters are used to design Audio digital filter [21]. Another application of Nyquist filter is frequency division multiplexing [22].

6. CONCLUSION

In this paper, work done on a Nyquist filter in the last decade has been discussed. In the last few years, there has been a great progress in the study of Nyquist filter. The Authors have proposed different methods to design Nyquist filter. The methods proposed in this, using different techniques are to improve the different parameters of the filter. The main work is done to get zero intersymbol interference, high stop band gain. Some of them have worked to improve power dissipation. While designing the Nyquist filter they have considered many factors and tried to improve the performance of this filter and some of its applications are also included in this review paper.

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Authors Biography



Ms. Kavita Rani has a background in Electronics and Communication engineering. She has completed her Bachelor of Technology (B. Tech) in Electronic and communication in the year 2014. Now, she is pursuing her Masters of Engineering (M.E) degree. Her research interests include communication, antenna design and its applications.



Mr. Deepak Saluja has a background in Electronics and Communication engineering. He obtained his Master of Engineering (M.E) degree in wireless communication from Thapar University, Patiala in 2015. He received his Bachelor of Technology (B. Tech) with Gold Medal in the year 2013. He has qualified GATE-2013 and 2016 in Electronics and Communication Engineering stream. He is having 3 years of experience in Teaching and Research. His research interests include wireless communication and signal processing.