International Journal for Research in Engineering Application & Management (IJREAM)
ISSN : 2454-9150 Vol-04, Issue-02, May 2018

A Survey on Non-Uniform Sampling using Level Crossing Techniques

Viswanadham R, Research Scholar, CUTM, Paralankhemundi, India, ravuri.viswanadh@gmail.com

S S Nayak, Professor, CUTM, Paralankhemundi, India, ssnayak@cutm.ac.in

Sudheer Kumar Terlapu, Associate Professor, SVECW, Bhimavaram, India, skterlapu@gmail.com

Abstract - Analog to Digital Conversion (ADC) is the foremost step in any digital communication application. Due to uniform sampling, traditional synchronous sampling systems causes high process complexity in further processing stages. Along with the complexity the power consumption is also more due to more number of computations. To avoid these problems non-uniform or asynchronous sampling techniques are introduced for time varying signals such as speech and audio signals. This paper presents a brief study on non-uniform sampling techniques Level-Crossing (LC) and Adaptive Level Crossing (ALC) schemes. An LC and ALC A/D converters are real-time asynchronous systems, which converts analog signal into a sequence of non-uniformly spaced time instants. The encoding and reconstruction process steps of LC and ALC schemes are presented in this paper along with the merits and demerits of the each scheme.

Keywords — ***asynchronous sampling, audio signals, encoding, level crossing, power consumption, reconstruction, uniform sampling.***

I. INTRODUCTION

Due to tremendous advancements took place in mobile and cellular communication, there is a need for high data rates with limited bandwidth. The performance of the communication systems is mainly depends on the utilization of the channel bandwidth with low power. The performance can also be increased by incorporating new methods of signal conversion techniques both at the transmitting end and at the receiving end. So, as to limit the bandwidth with low power consumption [1].

Bandwidth required for signal transmission is determined by the sampling rate of the signal. Uniform sampling requires more bandwidth and power [2].

In wireless sensor networks, it is important to reduce energy consumption of sensor nodes, most of the energy is consumed by data transmission, data acquisition due to more sampling rate. Therefore reduce the energy consumption it needs a new sampling technique, which adapts to time-varying frequency properties of the signal. This can be called signal-dependent sampling, when few samples are taken at low frequency regions and more samples at high frequency regions. The required bandwidth to transmit a signal depends on sampling rate and power requirement is estimated over total computations performed and number of samples transmitted [3].

The classical systems are based on the Nyquist signal processing architectures. These systems do not exploit the signal variations. Indeed, they sample the signal at a fixed rate without taking into account the intrinsic signal nature. It causes to capture, and to process a large number of samples without any relevant information, a useless increase of the system activity, and its power consumption. So uniform sampling is not preferable. By considering the non-uniform sampling need not to compute large number of samples, system activity and power consumption reduced. Obviously bandwidth required per user also decreased, channel utilization consumers number is increased [4].

Organization of this paper is as follows, Section II discuss about significant level crossing techniques and Section III about important adaptive level crossing techniques. In Section IV, techniques for reconstruction of signals are addressed, conclusions are drawn in Section V.

II. LEVEL CROSS SAMPLING TECHNIQUES

In non-uniform sampling, applied input signal is discretized using quantization at the transmission end and it is to be reconstructed as original signal at the receiver end. In this process quality of the signal should not be reduced. One basic technique is implemented to generate the samples by recording the different time instants at which the input crosses fixed quantized levels (Fig.1) [5]. Through employing various level crossings, requirements of time resolution can be reduced. This method generates samples of input signal that are non-uniformly sampled in time then it provides the series of amplitude –time pairs. If the quantized levels and non-uniformly spaced time instants are recorded to infinite precision then no error should be identified in this process [6].At every crossing, new sample is generated and average rate of samples is larger than the Nyquist rate. At the receiver end, uniform samples are required to generate the actual signal, so transformation of signal is essential to generate

 DOI : 10.18231/2454-9150.2018.0120

uniform samples from non-uniform samples. This transformation is possible by using infinite degree interpolation polynomial, it will calculate the amplitude value for any time instant. In this entire procedure, conversion of analog to digital (transmission side) and again analog signal is going to be generated (receiver side). So, A/D architecture is developed based on level crossing sampling (Fig.2). Here input signal is quantized to detect crossings at clock frequency (f_{clk}) . At every crossing new sample is generated that is applied to interpolator, it generates required uniform samples at f_{out}. Lastly, uniformed signal sequence is decimated to output in the chosen conversion rate, here picked to be the Nyquist rate. The rate at which the generation of uniform samples is limited by the speed of signal processing block as well as order of interpolation (accuracy).

At the input side of the level-crossing A/D converter, the applied signal is compared to a different quantization levels to identify the crossings. The accuracy of this process is depends on number of quantization levels, amplitude resolution, time resolution [7].

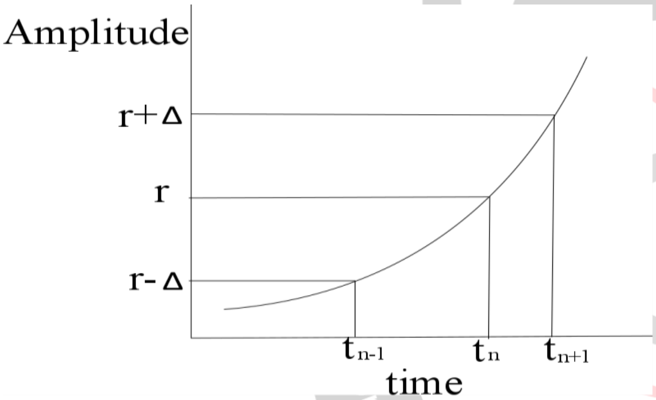


Fig. 1. Level-crossing sampling

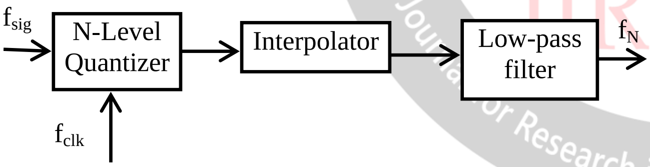


Fig. 2. Architecture of level-crossing sampling A/D converter

To apply the wide range of input signals, input sampling rate should be high and to achieve this periodic dither signal is added. For this new level crossing A/D implemented. Usually same process will be followed like previous technique but here triangular dither signal added to the input signal, the dither is assessed and vanished digitally before interpolation [8]. The architecture of ADC shown in Fig.3, it contains a dither generator and level-crossing detector in the input side (analog) and a dither estimator and signal interpolator in the digital side. The dither generator provides a triangle wave d(t), which is fed to the ADC input x(t).

The dither generator provides a triangle wave d(t), which is fed to the ADC input x(t).The resultant signal y(t) is generated by the level-crossing detector using comparators[9].

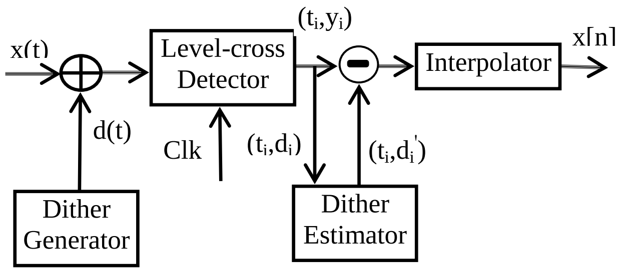
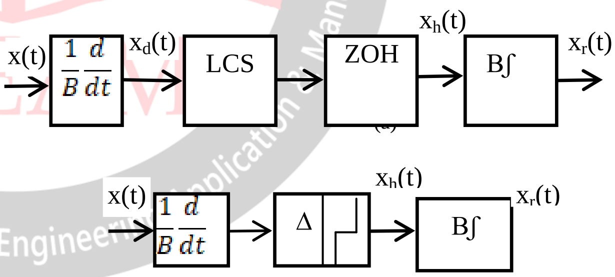


Fig. 3. ADC Architecture

Here the clock is provided with high speed for the purpose of quantization and to save the different time instants at which are levels crossed. The Level-cross detector provides the output as time-amplitude pairs as (t_i, y_i). For every pair, a digital estimate d_i (from d(t)) is shaped by the dither estimator and it is subtracted from y_i(level-crossing value) to z_i. To get the uniformed samples x[n],(t_i, z_i) pairs should be interpolated. These pairs represent the x(t) with a time period of T. In this technique there is no feedback then it avoids stability problems and there is no need to integrator (linear) but in sigma-delta converter having feedback. So this method of implementation is better than sigma-delta converter. The result of this A/D converter is over sampled; in this technique utilization of comparator is less [10].

The another approach is Derivative Level Cross Sampling, to improve the better performance by reducing the quantization error for bandpass signals At the transmitter end derivation operation is performed to the input signal then it is fed to the level cross sampling block, non-uniform samples are generated and transmitted and these samples are received at the receiver end and these are zero ordered, integrated then reconstructed linearly (piecewise) [11].



(b)

Fig. 4. (a) Principle of DLCS, (b) Equivalent system for analysis

Sparse signal reconstruction is also possible from the level crossed timings by using 1-bit compressive sensing model [12].Timing instants estimation in LCADC with respect to the input signal, those time instants are compared with conventional LC scheme and then error analysis also done based on irrelevant matching of time instants [13].Signals are encoded with LC scheme, based on variation of signal part with low error, fast varying part sampled fast and slow varying part sampled slowly [14].

 DOI : 10.18231/2454-9150.2018.0120

III. ADAPTIVE LEVEL CROSSING TECHNIQUES

A. Conventional Adaptive Level Crossing Technique

To obtain the better performance of LC converters new approach is described i.e. adaptive level crossing scheme. This method is also same as LC scheme but numbers of samples are increased with respect to the quantization levels based on number of bits/sec[15]. There is no possibility to identify the dynamic range of the applied signal to adjust the levels to overcome this problem ALC is suitable [16].Adaptive level crossing is choosen to obtain the accurate signal, in this samples are generated based on slope of the signal, step size, crossing time, frequency. From Fig.4, two levels are considered as q_1,q_2 and both are identified at low slope region or high slope region depends which level intersects the input signal.

If q1 met the signal first the, send a positive pulse then two levels are considered in the upstream and become q_1', q_2'. If q_2 met first then send a negative pulse and the two levels are progressed in the downstream. The values of levels obtained by these pulses. In this adaptive level crossing scheme, to adapt the samples four different cases are considered.

Case 1: Initially by considering the crossing levels as q_1, q_2. Whenever the applied signal intersects the crossing levels with positive slopes, the crossing levels will be updated as q_1', q_2' and low slope signal will be intersects first.

Case 2: whenever the applied input signal changes from positive region to negative region, the signal cannot be intersects with the upper crossing level.

Case 3: whenever the input signal meets crossing level with negative slope, the upper crossing level (q_2) will be met first.

Case 4: whenever the applied input signal changes from negative region to positive region, there is no possibility to intersect at least lower crossing level (q_1).

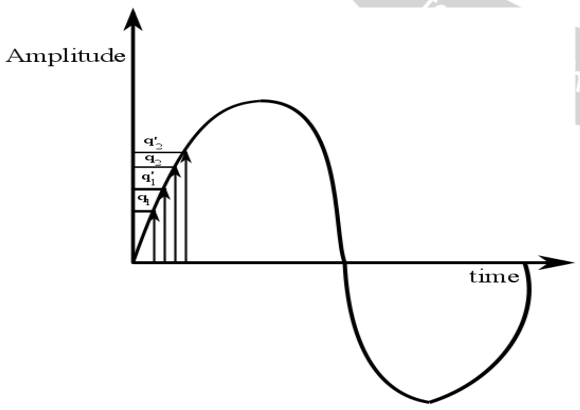


Fig. 5. Non uniform sampling based on Adaptive level crossing scheme

In LC scheme, range of the input signal (dynamic range) is required to set right the crossing levels. ALC scheme follows the adaptive method to input signal without knowing the dynamic range. In these level crossing schemes space (distance) between crossing levels is same and they will not provide more number of samples where input signal slope is low. To increase the more number of samples new solution is described i.e. multilevel ALC, here space between levels is adapted to the slope of the input signal, this solution provides the enough number of samples especially in low slope regions of the applied signal. Whenever the number of samples increased obviously performance of the converters is also improved [17].

B. Flexible Adaptive Level Crossing Architecture

The applied input signal divided into two different parts one is active section (time period, is having more number of amplitude variations), the other is inactive section (time period, is having less number of amplitude variations) [18]. From Fig.6.a. is a sampled uniform signal because time space between any two samples is same. From Fig.6.b. is sampled signal with level crossings, distance between any two samples is not same and from the both plots may observe the signal amplitude variations as active section and inactive section. This method permits the consumers to adjust three important considerations i.e. highest quantization step, lowest quantization step and no-crossing time. The highest and lowest quantization steps are the maximum and the minimum quantized step values that can be done by QSA algorithm. Whenever the signal passes without crossing the levels (fixed), maximum time required to this process is called no crossing time.

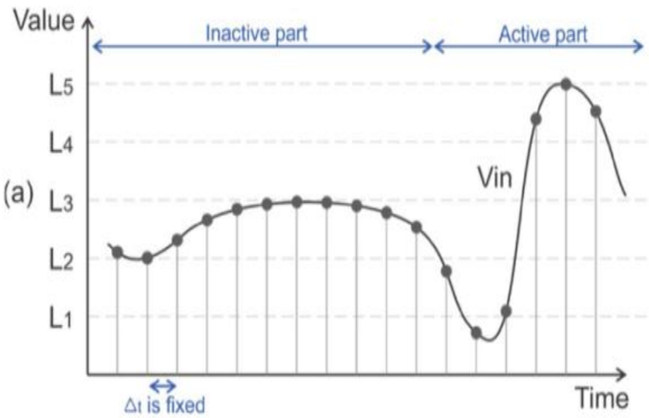


Fig. 6 (a). Sampled Uniformed Signal

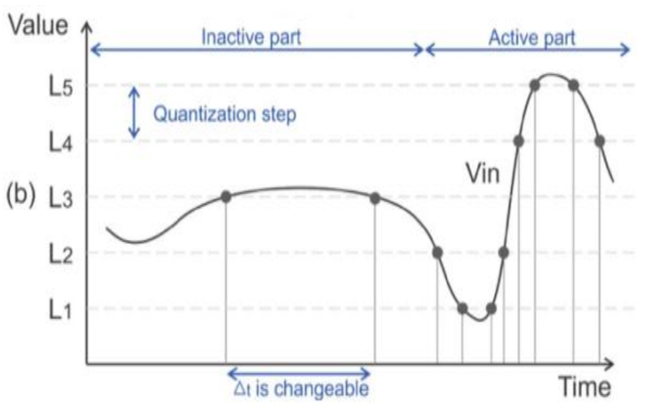


Fig. 6 (b). Sampled Signal received with crossings

 DOI : 10.18231/2454-9150.2018.0120

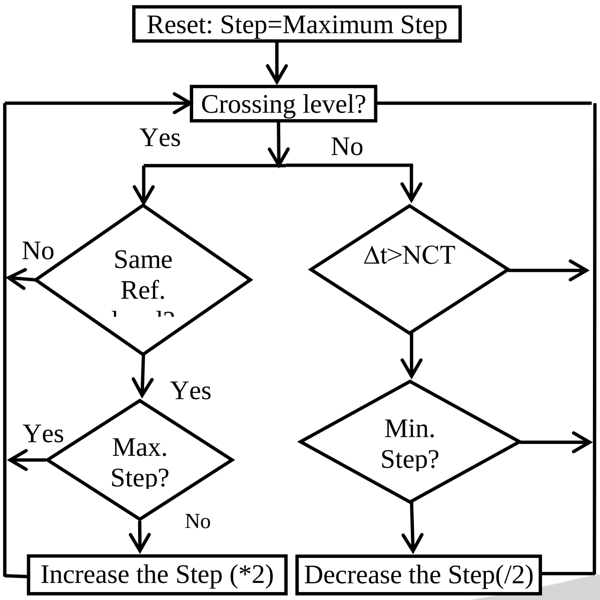


Fig. 7. QSA Algorithm

If the no crossing is identified, converter upgrades the quantization step. For every crossing, quantization step is doubled each time; otherwise there is no change in quantization step. If the no crossing time is less than the Δt then the quantization step is decreased [19].

C. Linear Adaptive Level-Based Sampling

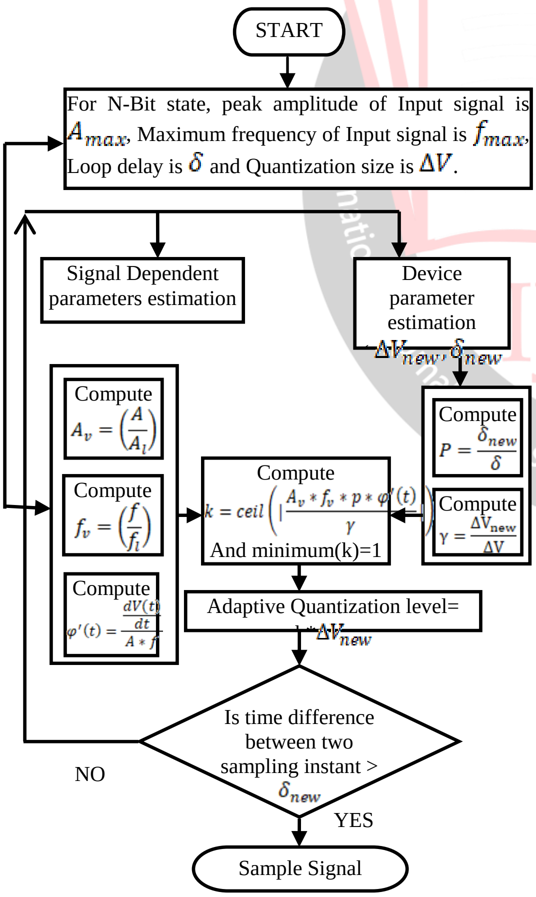


Fig. 8. LALBS Algorithm

In general level based sampling, rate of change of sampling depends on how fast signal crosses the quantization levels and it is controlled by system loop delay. This loop delay determined as minimum time required to complete one conversion. From the below flowchart (Fig.8), change in the value quantization leads to change in the value of K and there is possibility to measure the loop delay when K>1, it assures time difference between the samples is equal to loop delay. Let total delay of ADC, DAC be δ, if any change in input loop delay will changed to δnew. The adaptive reference level is completely depends on the k value and ΔVnew (modified quantization of ADC). These are the some important adaptive level crossing techniques to obtain the more number of samples to reconstruct the accurate applied signal [20].

IV. Reconstruction Techniques

Based on the transformation technique may introduce extra errors, those will limit the overall resolution. Truncation errors are depends on the order of interpolation polynomial. Because of uncertainties in time and amplitude values of different samples may cause rounding errors. By increasing the order of interpolation and number of quantization levels, SNR will be improved. In practical A/D converters, SNR levels increased by using second order interpolator. Decimator will guess the entire performance A/D converter for a given parameters. Decimator is used to filter the noise then the converter resolution is increased [21].

A. Using G Operator

Reconstruction of the signal from non-uniform samples and quality of the signal can be succeeded by iterative algorithm. From Fig.9 G operator is used to reconstruct the original signal at the receiver end. Non uniform samples are interpolated linearly then it is applied to the low-pass filter. To obtain the much quality of the reconstructed signal number of iterations should be increased [22].

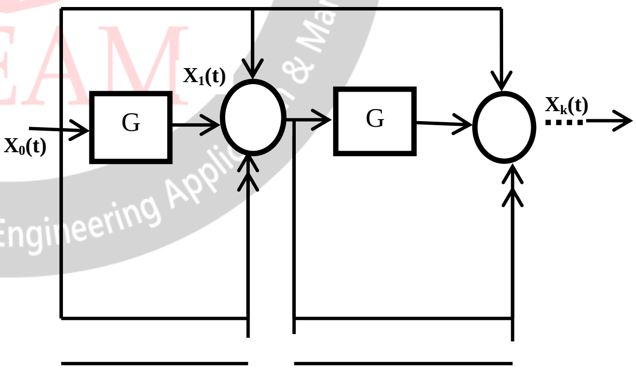


Fig. 9. Iterative Operation

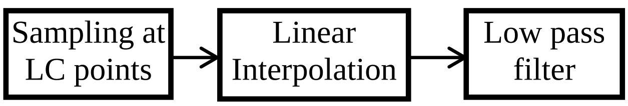


Fig. 10. G operator in Iterative Operation

B. Using IASR Algorithm

The below block diagram of IASR referred as Bandlimited Interpolation Approximation (BIA), initially it performs a bandlimited interpolation of h parametrized by Δ(hΔ(u)) and it is processed by M1/α.Reconstruction of bandlimited signal

 DOI : 10.18231/2454-9150.2018.0120

using IASR is better than varnoi method because in terms of rate of convergence when the sampling density approaches the Landau rate[23].

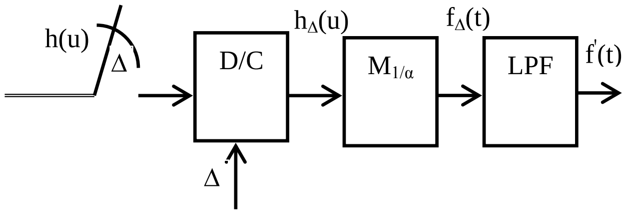


Fig. 11. IASR Algorithm

V. CONCLUSION

This paper presented a brief literature survey on various asynchronous sampling techniques such as LC and ALC techniques. The benefits of these techniques applicable only for non-stationary or time varying signals. If the signal is stationary then resultant complexity with non-uniform sampling techniques are equal to uniform sampling techniques. Among LC and ALC, LC has simple to implement but the reconstruction of the signal is difficult when the step size is more. Whereas ALC is difficult to implement but the reconstruction gives exact original signal.

REFERENCES

[1] MARK, J.W., and TODD, T.D. : "A nonuniform sampling approach to data compression", IEEE Transactions on Communications, Vol. COM-29, No.1, Jan. 1981, pp. 24-32.

[2] F. Marvasti, Nonuniform Sampling Theory and Practice. New York: Kluwer Academic, 2001.

[3] Mahdi Boloursaz Mashhadi, Student Member, IEEE, and Farokh Marvasti, Senior Member, IEEE,"Iterative Methods for Sparse Signal Reconstruction from Level Crossings", arXiv:1611.10136v1 [cs.IT] 30 Nov 2016.

[4] Nassim Ravanshad, Hamidreza Rezaee-Dehsorkh and Reza Lotfi, "Detailed Study of the Time Estimation in Level-Crossing Analog-to-Digital Converters", 2013.

[5] N. Sayiner, H. V. Sorensen, and T. R. Viswanathan, "A new signal acquisition technique," in Proc. 35th Midwest Symp. Circuits Syst., Wash., DC, Aug. 1992, pp. 1140-1142. M. Young, The Technical Writer's Handbook. Mill Valley, CA: University Science, 1989.

[6] N. Sayiner, H. V. Sorensen, and T. R. Viswanathan, "A Level-Crossing Sampling Scheme for A/D Conversion", IEEE transactions on circuits and systems-11: analog and digital signal processing, Vol. 43, No. 4, April 1996.

[7] P. W. Jungwirth and A. D. Poularikas, "Improved Sayiner level crossing ADC," in Proceedings 36th Southeastern Symp. Syst. Theory, Mar.2004, pp.379–383.

[8] Tünde Wang, Dong Wang, Paul J. Hurst,Bernard C. Levy, Stephen H. Lewis, "A Level-Crossing Analog-to-Digital Converter with Triangular Dither", IEEE transactions on circuits and systems—i: regular papers, vol. 56, no. 9, September, 2009.

[9] Kafashan, M., Beygiharchegani, S., Marvasti, F.,Performance improvement of an optimal family of exponentially accurate sigma delta modulator'. 10th IEEE Int.Conf. on Signal Processing (ICSP), 2010,pp. 1–4.

[10] Pablo Martınez-Nuevo, Sharvil Patil, Yannis Tsividis, "Derivative Level-Crossing Sampling",IEEE Transactions on Circuits and Systems II:Express Briefs, 2015.

[11] V.Silva, A.L. Souza and S.Catunda, "Flexible A/D converter architecture targeting sparse signals",2014 IEEE International Instrumentation and Measurement Technology Conference(IIIMTC)Proceedings, Montevideo, 2014, pp. 1496-1500.

[12] Ravanshad, Nassim, Hamidreza Rezaee-Dehsorkh, and Reza Lotfi. "Detailed study of the time estimation in level-crossing analog-to digital converters", 2013 21st Iranian Conference on Electrical Engineering (ICEE),2013.

[13] Tsividis Y. Event-driven data acquisition and digital signal processing a tutorial. IEEE Transactions on Circuits and System II: Express Briefs 2010; 57(8).

[14] Nematollah Zarmehi, Sina Shahsavari and Farokh Marvasti, "Comparison of uniform and random sampling for speech and music signals", Sampling Theory and Applications (SampTA),IEEE publications,Estonia,2017.

[15] Karen M. Guan, Suleyman S. Kozat and Andrew C.Singer, "Adaptive Reference Levels in a Level- Crossing Analog-to-Digital Converter", EURASIP Journal on Advances in Signal Processing, Volume 2008, Article ID 513706.

[16] Mehrzad Malmirchegini, Mohammad Mehdi Mona Ghassemian, Farokh Marvasti, "Non-uniform Sampling based on an adaptive level-crossing Scheme", IET Signal Process. 2015, Vol. 9, Iss. 6, pp. 484–490.

[17] Verônica Maria, Antonio Augusto, Lisboa de Souza, Sebastian Yuri avalcanti, Raimundo Carlos Silvério Freire, "Non-Uniform Sampling based ADC Architecture using an adaptive level-crossing technique", IEEE transaction, 2007.

[18] R. Viswanadham, T. Sudheer Kumar, M.Venkata Subbarao. "A Level cross-Based Nonuniform Sampling for Mobile Applications", Springer Nature, 2018, Chapter 79.

[19] Abhinav Srivastava, K. Srinivasan, Development of mathematical model for signal-dependent linear adaptive level-based sampling, Int. J. Circ. Theory.Appl. 2016; 44:1112–1126, published online 8,September 2015 in Wiley Online Library.

[20] Greitans, M.: 'Processing of non-stationary signal using level-crossing sampling'. Proc. SIGMAP, August 2006, pp. 170–177.

[21] F. Marvasti and T. Lee, "Analysis and recovery of sample-and hold and linearly interpolated signals with irregular samples," IEEE Transactions in Signal Process., vol. 40, no. 8, pp.1884 –1891, Aug. 1992.

[22] M. Malmir Chegini and F. Marvasti,"Performance Improvement of Level-Crossing A/D Converters",IEEE international conference on communications,Penang, Malaysia, May 2007.

[23] Hsin-Yu Lai, Pablo Martınez-Nuevo, and Alan V.Oppenheim, "An Iterative Reconstruction algorithm for Amplitude Sampling", ICASSP 2017 @ IEEE.

 DOI : 10.18231/2454-9150.2018.0120