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A NOVEL LOW LATENCY, HIGH RESOLUTION AND LOW COST TIME SYNCHRONIZATION

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A NOVEL LOW LATENCY, HIGH RESOLUTION AND LOW COST TIME SYNCHRONIZATION

By

Ali Aghdaei

A REPORT

Submitted in partial fulfillment of the requirements for the degree of

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In Electrical Engineering

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This report has been approved in partial fulfillment of the requirements for the Degree of MASTER OF SCIENCE in Electrical Engineering.

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Dedication

This work is dedicated to to my parents and my sister

for their endless love, support and encouragement

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Abstract

This report presents a new low latency, high resolution and low cost timing synchronization technique for digital receivers. Traditional timing synchronization employs Matched filter to perform cross-correlation operation and estimate Time-of-Arrival (TOA) of the signal. Decreasing the latency of the traditional method through oversampling leads to a higher complexity and it is not viable. Furthermore, to obtain a high-resolution TOA, an extensive bandwidth is required, which results in high system complexity. The proposed method uses single bit quantization to employ XNOR blocks instead of multiplier and accumulator (MAC) blocks in the traditional method. This substantially decreases complexity incorporating less hardware elements in FPGA surface area.

Chapter 1

Introduction

There is a broad range of applications for localization and researchers have offered many technologies to improve performance in various aspects[1, 2, 3]. Examples of localization applications are: battlefield surveillance [4], body health monitoring[5], automated vehicles [6], border monitoring[7] and law enforcement[8]. For emergency 911 services, it is vital to locate the exact position of the user and offer support in the shortest time period[9]. Time-of-Arrival (TOA) estimation is a popular method for localization [10] and time synchronization is the coarse component of TOA estimation [3]. Improving performance, cost and latency of coarse TOA estimation greatly impacts the operation of all receivers in general and localization techniques in particular.

TOA estimation includes two stages of coarse and fine[11]. Matched filters are traditionally used for receiver time synchronization or coarse TOA estimation[12]. Examples of fine TOA methods include multiple signal classification (MUSIC) [13], Blind Signal Separation (BSS) [10] and Independent Component Analysis (ICA)[14]. In general, Matched filters suffer from high latency, low resolution and high complexity[15]. In[16], a TOA architecture is designed to reduce the number of hardware elements in Matched filters. However, the designed system doesn't address the conversion factor. Conversion factor represents a scale to convert a decimal number to a floating-point binary format. It plays a crucial role in the complexity evaluation of the system. Thus, the complexity of the system is still high when high resolution TOA estimation is needed. In[17], a time synchronization method which employs oversampling is proposed to increase the performance of the system. However, the probability of error is increased applying oversampling technique. It is also cost inefficient as higher number of hardware blocks are needed to perform TOA estimation. Authors in [18],[19] investigate hardware design to increase the area efficiency, but the reported results are specific implementation without a generic cost function.

This report proposes a TOA measurement method that improves resolution via oversampling technique. The oversampled signal is split into distinguished paths to overcome low time synchronization reliability associated with oversampling process. TOA of the signal is attained with higher resolution based on the combination of TOA estimations obtained from different paths. To address the complexity issue associated with oversampling, XNOR blocks are used. Here, bit conversion is applied to allow using XNOR blocks. Performance-complexity trade off as a function of oversampling ratio and word-length is investigated. It should be noted that oversampling could reduce the latency of both Matched filter and the proposed approach. However, since oversampling highly increases the complexity of Matched filter, lower latency would not be practically feasible.

The report is organized as follows: Chapter 2 introduces the proposed time synchronization method, and discusses the sources of error, TOA parameters and measures to evaluate the performance. In Chapter 3 both methods are mathematically evaluated. Chapter 4 compares the performance and cost of the proposed method with the traditional Matched filter method. Chapter 5 concludes the report.

Chapter 2

The Proposed Time Synchronization

Latency, resolution and complexity are significant measures which indicate the superiority or inferiority of a time synchronization method. The received signal delay is random due to the channel propagation and device random delays. Device delay is a non-deterministic error that is created by traditional Matched filter time synchronization process[20]. Resolution is a function of sampling time (T_s), signal-to-noise ratio (SNR) and the pilot size[21]. Throughout the report, we call the pilot size as word-length. Decreasing T_s increases the bandwidth that subsequently increases the resolution. Thus, T_s represents the minimum attainable resolution through time synchronization process[22]. In addition, it is observed that increasing the word-length in the pilot decreases system delay and improves TOA resolution [23]. However, increasing the word-length increases system complexity. Since all digital receivers are limited in the surface and cost, designing a low complexity time synchronization system is desirable. Thus, it is necessary to select a proper trade-off between the accuracy and complexity based on restriction imposed by applications. Main factors that impact complexity include word-length and conversion factor (C_F). Transmitting a larger word-length requires more hardware and occupies greater surface on FPGAs. To implement TOA estimation on FPGA, we should convert floating point to binary format. Depending on the required accuracy in TOA, C_F is assigned a different value. For example, more number of multiplier blocks is required to perform a multiplication operation in 32-bit format compared to 8-bit format.

2.1 Matched Filter Synchronization

In order to assess the traditional Matched filter synchronization, the transceiver model of Fig.2.1 is used. Here, the the channel is considered flat with the impulse response of:

$$h(t) = \alpha\delta(t - T) \tag{2.1}$$

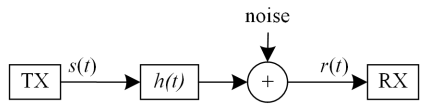

Figure 2.1: Transceiver Model

In (2.1), α is a complex random variable, $\delta(.)$ is Dirac delta function and T is the time of arrival. Thus, the received signal corresponds to :

$$r(t) = \alpha s(t - T) + n(t) \tag{2.2}$$

where $s(t)$, $0 < t < T_o$ represents the transmitted waveform with the duration of T_o and $n(t)$ is white Gaussian noise. Traditional Matched filter TOA estimation attempts to calculate T. It works based on the cross-correlation between the received signal and a known template signal stored in the receiver that is represented as:

$$Y[n] = \sum_{m=0}^{N-1} P[m]r[n + m] \tag{2.3}$$

where $r[n]$ and $Y[n]$, $n = 1, 2, ..., N$ are the sampled received signal using T_s and the output value (Matched filter output) at $t = kT_s$, $k = 1, 2, 3, ...$ respectively. N is the word-length of pilot and P is the template waveform. TOA is the earliest arrival time that maximizes the cross correlation operation of (2.3). It is designed with a non-coherent correlator followed by a low-pass filter shown in Fig.2.2.

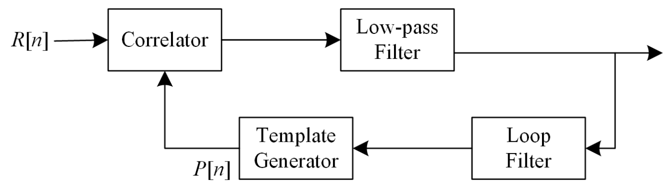

Figure 2.2: Matched filter block diagram

Matched filter performance varies with bandwidth and increasing the bandwidth increases the complexity. The loop shown in Fig.2.2 produces a non-deterministic latency in Matched filter system[24].

2.2 Proposed Method

In the proposed method, oversampling is employed to obtain higher TOA resolution. It keeps the bandwidth intact since oversampling is just applied to the receiver side. Thus, the complexity related to enlarging bandwidth does not impact the proposed TOA method.

Oversampling leads to two issues in the receiver. First, it minimizes the difference between the output of Matched filter and its side lobes. Thus, the possibility of error in detecting the maximum point of the cross-correlation function increases. This causes an error in TOA estimation which makes the time synchronization unreliable.

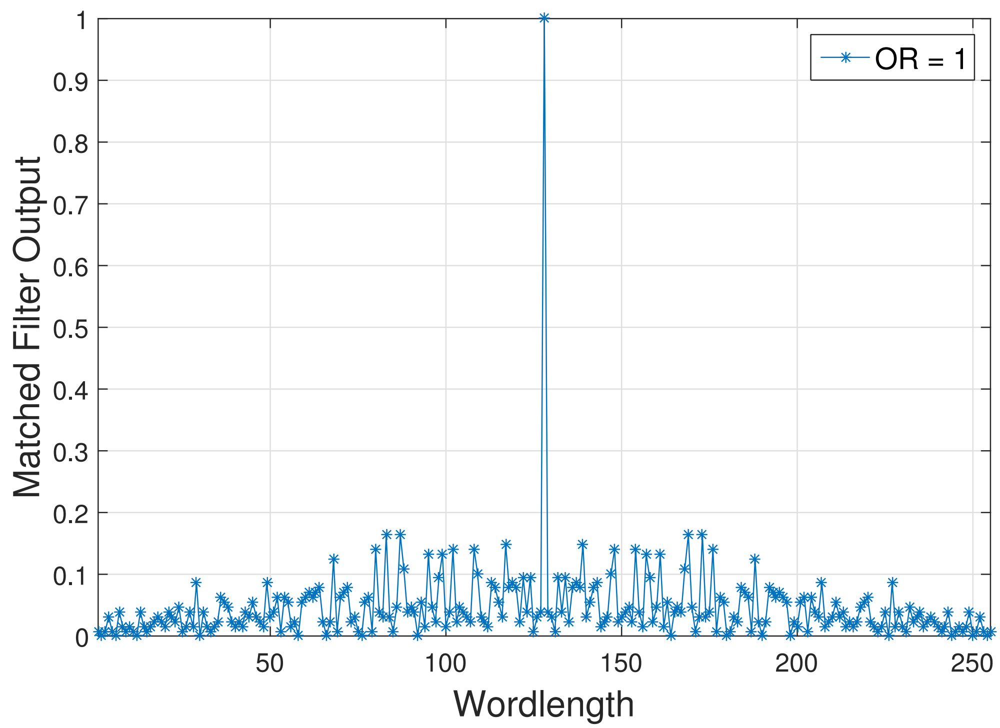

Figure 2.3: Matched filter output (OR = 1)

Fig. 2.3 represents the out output of Matched filter when OR = 1. It is observed that the difference between peak and the next greatest value is about 0.8. In Fig2.4, Matched filter output at OR = 4 is illustrated. It is noticed that the difference between the peak and the next greatest value is 0.2. Thus, the probability of error to discover the peak when OR = 4 is increased in the presence of noise.

Increasing system complexity is another negative impact of oversampling. When the received signal is oversampled with a higher ratio, more number of operations are needed to perform. This leads to a cost-inefficient system even if high TOA resolution

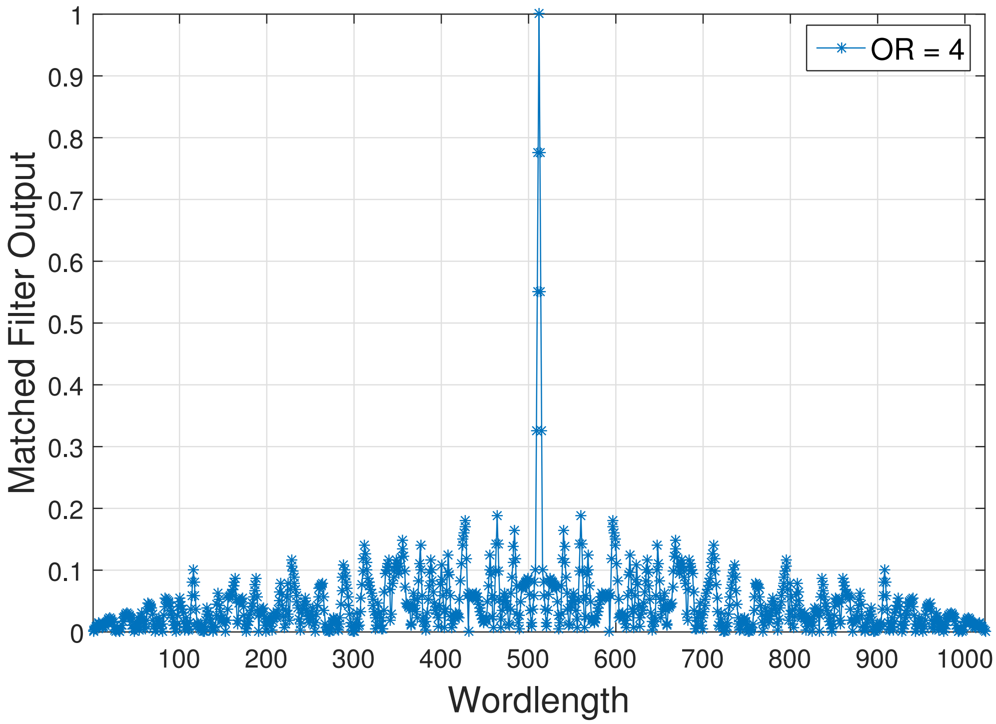

Figure 2.4: Matched filter output (OR = 4)

is achieved.

To approach the first issue, the oversampled received signal is split into L discrete paths where $L = F_s/F_N$ where F_s and F_N are the sampling frequency and Nyquist frequency of the received signal respectively. From (2.3), the received signals over paths 1 to L, i.e., $Y^{(1)}[n] - Y^{(L)}[n]$ correspond to:

$$Y^{(1)}[n] = \sum_{m=0}^{N-1} P[m]r^{(1)}[n+m]$$

$$\vdots$$

$$Y^{(2)}[n] = \sum_{m=0}^{N-1} P[m] r^{(2)}[n+m]$$

$$Y^{(L)}[n] = \sum_{m=0}^{N-1} P[m] r^{(L)}[n+m] \tag{2.4}$$

Therefore, L different values are attained over each path. These values are applied to decision block to determine TOA of the signal. Now, TOA is estimated when $Y^{(1)}[n]$ - $Y^{(L)}[n]$ are maximized. Therefore, TOA is decided at $t = k\hat{T}_s$ where $\hat{T}_s = T_s/L$. Thus, higher resolution TOA estimation is achieved due to reducing T_s.

This method solves the issue of increasing the probability of error in detecting the maximized output of the correlation. This algorithm also increases TOA resolution L times greater compared to that of non-oversampling.

Traditional Matched filter TOA method tends to be inefficient. It entirely relies on multiplier and accumulator (MAC) blocks to perform cross-correlation function. MAC blocks are cost inefficient due to necessity of large amount of hardware elements to execute the operation. They also occupy large area on FPGA surface, which is undesirable. As mentioned, oversampling increases the complexity of the system. For the proposed method, we apply XNOR operation instead of correlation to reduce system complexity.

To make XNOR operation feasible, a sign function is applied to the received samples.

This converts the floating point values to the binary format. This operation decreases system complexity in two separate phases. First, using XNOR blocks instead of MAC blocks that occupies more surface on FPGA leads to reduced system complexity. Second, due to applying sign function to the collected samples, the conversion factor (C_F) is decreased to 1. It should be noted that in this method, the hardware required to perform a multiplication between two 8-bit float numbers is replaced by hardware required to perform XNOR function between two 1-bit numbers. This, substantially increases the cost efficiency of the proposed TOA method compared to the traditional method. The system model of the proposed method is illustrated in Fig.2.5

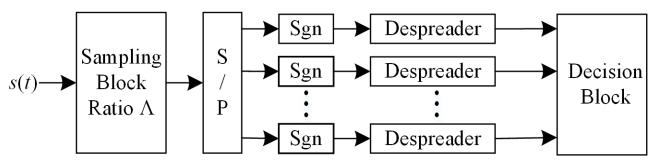

Figure 2.5: System Model

In Fig.2.5, de-spreader block is responsible for performing XNOR operation. Fig.2.6 represents the details of de-spreader block. In addition, this process leads to a low latency in the operation in the Matched Filter loop of Fig.2.2. It should be noted that oversampling could reduce the latency of both Matched filter and the proposed approach. However, since oversampling highly increases the complexity of Matched filter, then lower latency would not be practically feasible.

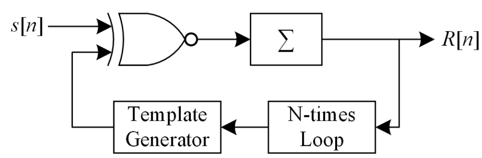

Figure 2.6: The Proposed Block Diagram Using XNOR

Chapter 3

Mathematical Evaluation

Here, we study the performance and cost of the proposed technique.

3.1 Performance

We assume $\hat{T}$ is the estimation of the time delay (T) introduced in (2.2) and defined as:

$$\hat{T} = T + \Delta \tag{3.1}$$

Here, Δ is an independent zero mean random variable. Mean square error (MSE) of the estimated T corresponds to:

$$MSE(\Delta) = E[(\hat{T} - T)^2] \tag{3.2}$$

by adding and subtracting $E(\hat{T})$ in (3.2), we have:

$$MSE(\Delta) = E[(\hat{T} - E(\hat{T}) + E(\hat{T}) - T)^2] \tag{3.3}$$

expanding (3.3), and given that Δ is zero mean, T is deterministic and $E(\hat{T}) - T = 0$, we have:

$$MSE(\Delta) = E[\hat{T} - E(\hat{T})]^2 = \sigma_{\hat{T}}^2 \tag{3.4}$$

where $\sigma_{\hat{T}}^2$ is the variance of $\hat{T}$.

It is proved that Cramer-Rao bound of variance of time delay corresponds to [25, 26, 27]:

$$\sigma_{\hat{T}}^2 \geq \frac{1}{A^2 B^2} \tag{3.5}$$

where

$$A^2 = \frac{2E}{N_o} \tag{3.6}$$

and

$$B^2 = \frac{\int_{-\infty}^{+\infty} \omega^2 |F(\omega)|^2 d\omega}{\int_{-\infty}^{+\infty} |F(\omega)|^2 d\omega} \tag{3.7}$$

Here, E is the energy of the transmitted signal $s(t)$, $F(\omega) = \int_{-\infty}^{+\infty} s(t)e^{-j\omega t}dt$ and B is a measure of bandwidth.

Assuming that the signal spectrum is two sided between f_1 to f_2 HZ with spectral density of $s_o/2$, we have:

$$\begin{aligned} B^2 &= \frac{2\int_{f_1}^{f_2} (2\pi f)^2 (2\pi s_o/2) df}{2\int_{f_1}^{f_2} (2\pi s_o/2) df} \\ &= (2\pi)^2 (f_2^2 + f_1 f_2 + f_1^2)/3 \end{aligned} \tag{3.8}$$

Applying (3.6) and (3.8) into (3.5), we have:

$$\begin{aligned} \sigma_{\hat{T}}^2 &\geq \frac{1}{\frac{2E}{N_o}\frac{4\pi^2}{3}(f_2^2 + f_1 f_2 + f_1^2)} \\ &= \frac{1}{\frac{2sC}{N_o(f_2-f_1)}\frac{4\pi^2}{3}(f_2 - f_1)(f_2^2 + f_1 f_2 + f_1^2)} \end{aligned} \tag{3.9}$$

Here, $C = N \times T_s$, where N and T_s are word-length and sampling time, respectively. Considering $SNR = s/N_o$ and substituting the defined parameters in (3.9), we have:

$$\sigma_{\hat{T}}^2 \geq \frac{3}{8\pi^2 N T_s} \frac{1}{SNR} \frac{1}{(f_2^3 - f_1^3)} \tag{3.10}$$

Based on (3.10), it is concluded that there is an inverse relationship between $\sigma^2_{\hat{T}}$ and signal-to-noise ratio (SNR), sampling time (T_s) and word-length (N). The results in (3.10) offers a lower limit for MSE. In Chapter 4, we observe that MSE performance of the proposed method is much closer to this lower limit.

3.2 Complexity

To evaluate the complexity of the system, the number of hardware elements used in each algorithm is considered as the crucial parameter. This parameter is used to mathematically compare the traditional Matched filter and proposed method. In Fig. 3.1, the complexity of the system is investigated considering cross-correlation function between the received samples and template.

In Fig.3.1, r_n and P_n are the received samples and the template respectively. N is word-length and C_F is conversion factor. In the traditional TOA algorithm, the operation between P_n and r_n is multiplication, which increases system complexity due to high value of C_F. The number of operations performed to obtain TOA is C_F^2. Thus in the traditional Matched filter, in case of $C_F = 8$, 64 operations are completed to estimate TOA. However, in the proposed method C_F is set to 1 due to applying XNOR blocks in the architecture. Therefore, the contribution of XNOR blocks decreases the complexity of the traditional Matched filter system from NC_f^2

to N in the proposed method which is a great improvement.

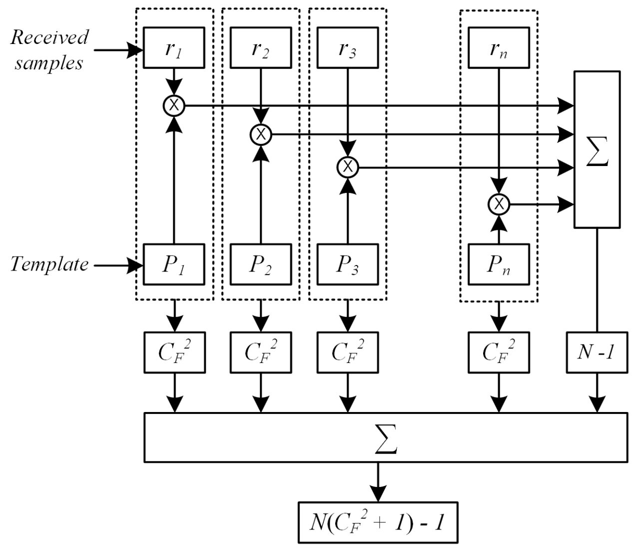

Figure 3.1: TOA Algorithm Architecture

To compare the system complexity in traditional Matched filter and the proposed method in a fare condition, both systems are implemented employing NAND gates. The number of NAND gates is required to implement AND, OR and XNOR functions are 2, 3 and 5 respectively[28].

Chapter 4

Simulations and Discussions

Simulations are conducted to evaluate the proposed TOA technique resolution and cost. The simulations assume BPSK modulation, flat fading channel, sampling rate of 1MHz, and a random sequence with word-length $N = 128$. In the traditional Matched filter, the convolution operation is performed between the template (original transmitted sequence) and the received samples. Whereas, in the proposed method XNOR operation is applied. MSE is used to evaluate the error imposed on TOA estimation in various SNR values. The received signal is oversampled with various ratios through decreasing the sampling time (T_s).

Fig.4.1 compares the proposed and traditional methods in terms of resolution with respect to complexity. SNR is set to 5 and word-length is considered 64. The criteria for complexity is the number of NAND gates used to perform TOA estimation. MSE is calculated for both methods at the same hardware complexity. A great improvement in TOA resolution of the proposed method compared to the traditional method is achievable. It is seen that at the complexity of 4000, MSE of the proposed method is about 100 times less than that of traditional method. The same improvement scale is also visible for different values of complexity. Theoretical results are also compared to both methods. Results justify that the implementation of the proposed method leads to results closer to the ideal Cramer-Rao MSE.

Fig.4.2 represents the performance of the traditional and proposed TOA with different word-length at different SNRs. Oversampling ratio is constant and equals 2. In this figure, T and P stand for traditional and proposed method, respectively. The simulation is repeated for different word-length (N). It observed that better performance is achievable using the proposed method compared to the traditional Matched filter method at each word-length. This is due to the contribution of XNOR blocks which decreases the possibility of detecting error in the proposed TOA compared to Matched filter. Furthermore, it is seen that the performance of both methods is improved by increasing word-length (N). However, this is not practically feasible in the traditional Matched filter method due to increasing complexity.

Simulations in Fig.4.2 were conducted assuming flat fading channel. However, many realistic channels (specifically at higher bandwidth) are Frequency selective. To address inter-symbol interference that is experienced in frequency selective channels, orthogonal frequency multiplexing (OFDM) is used.

Fig.4.3 considers a 3 tab power delay profile for the channel and OFDM transmission with 128 parallel channels. The results confirm that better performance in both systems is achievable increasing oversampling ratio (OR). This is due to the decreased sampling time which increases the resolution. Moreover, the performance of the proposed method is improved as OR increases. This is the result of XNOR blocks contribution since the probability of a wrong TOA estimation is reduced in each parallel path of Fig.2.5.

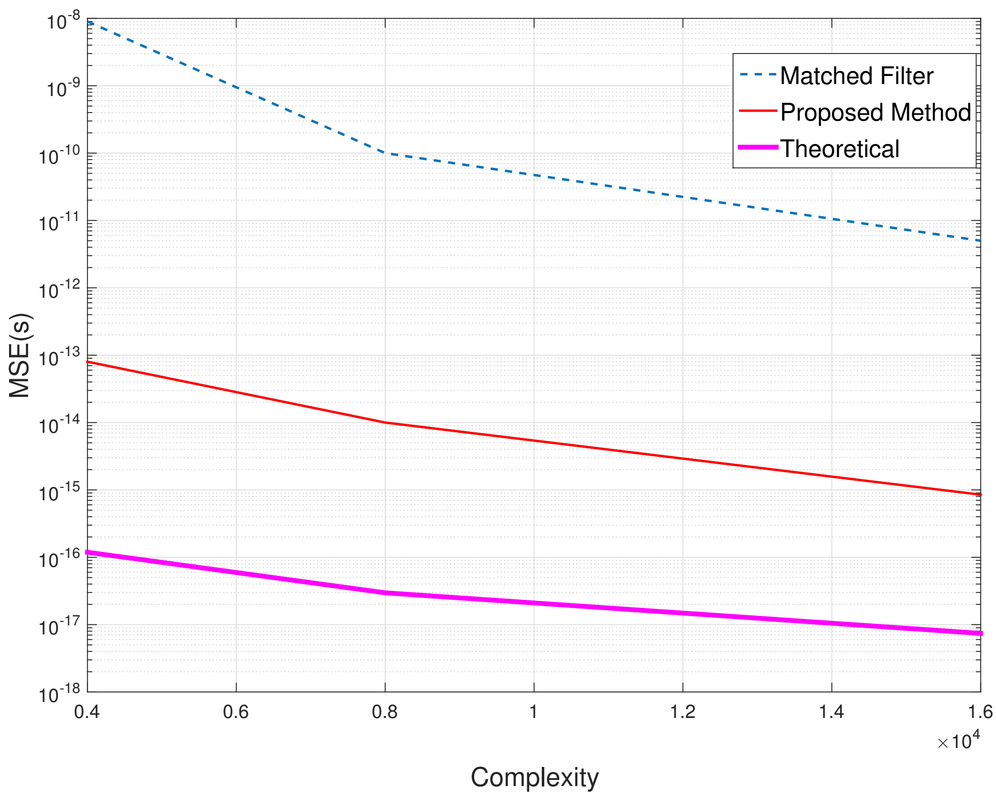

Figure 4.1: Complexity Vs Performance of the traditional Matched filter and Proposed methods

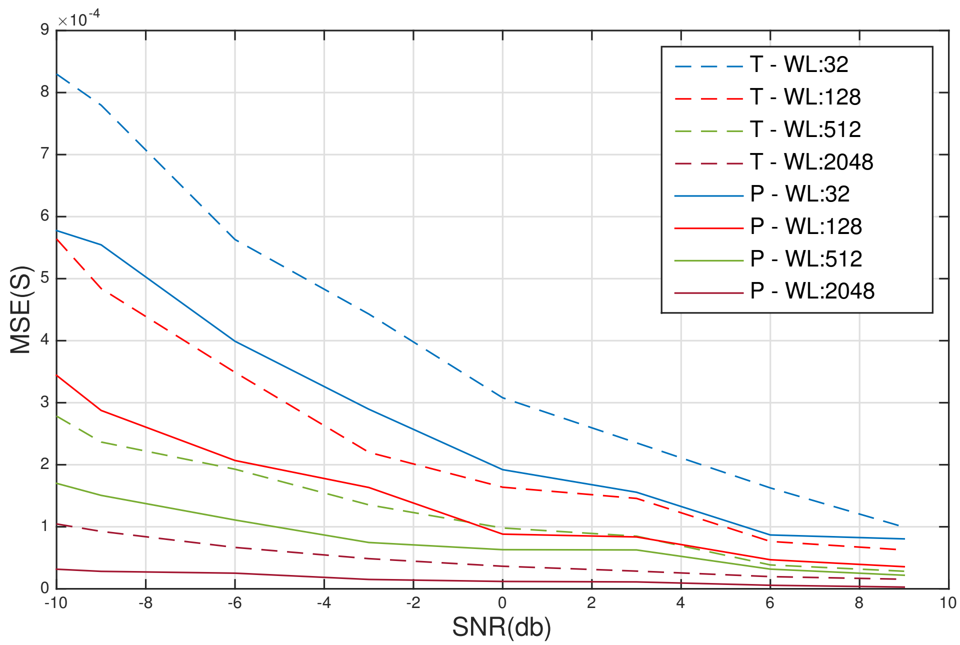

Figure 4.2: Performance of The Traditional and Proposed methods in different SNR Values

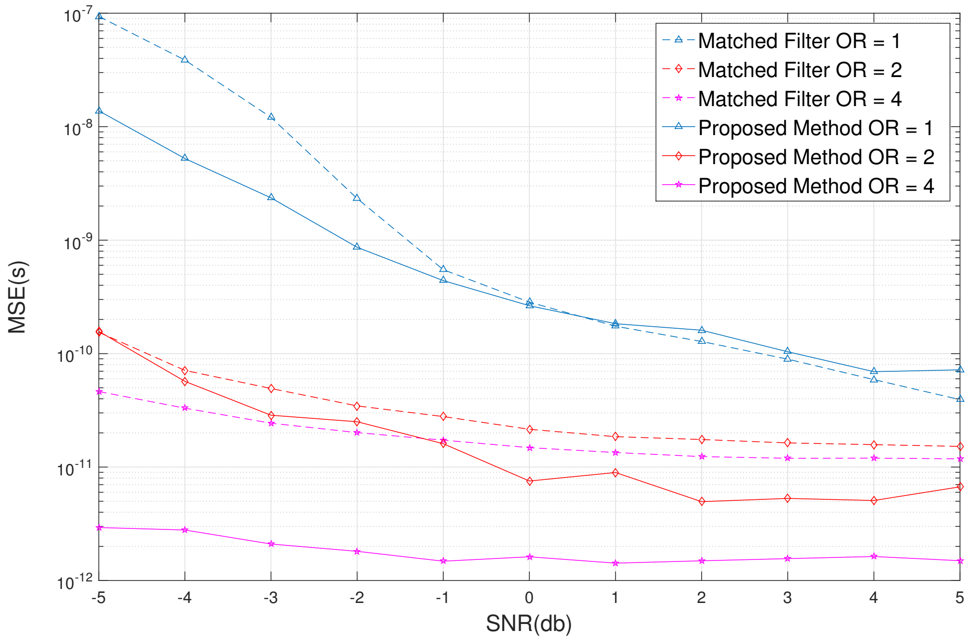

Figure 4.3: Performance of The Traditional and Proposed methods with different over sampling ratios (OR) using OFDM technique

Chapter 5

CONCLUSION

This report presents a novel low latency, low cost and high performance time synchronization method for digital receivers in general and localization in particular. The complexity and performance evaluation confirms that the proposed time synchronization method delivers a better performance compared to the traditional Matched filter method. It is also determined that the complexity of the system is reduced. This allows the implementation of time synchronization on a smaller FPGA surface while achieving a higher performance and lower latency. This enables the development of high performance, low cost wireless nodes for numerous sensor network applications that require both communication and localization. Examples are vehicle-to-vehicle (V2V) joint communication and localization, collaborative driving and environmental sensing.

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