ESTIMATION OF SYMBOL TIMING AND CARRIER FREQUENCY OFFSET USING SYNCHRONIZATION SCHEME

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ABSTRACT

OFDM/OQAM is preferred as multicarrier system which operates over a multipath channel. By using the multipath channel the signal-to-noise ratio. In earlier, sub carriers are used to transmit the signals. Nowadays, FFT and DFT are used for transmitting the signals based upon the bit values. AWGN is a channel used to identify the noise produced at the output by adding the noise in the blind signal. By reducing subcarriers the noise and timing are reduced. FFT bit value was increased which provides better performance. In the multicarrier system, the error and noise was reduced by increasing the bit value.

KEYWORDS: *OFDM, symbol timing, cyclic prefix, AWGN, signal-to-noise ratio, carrier frequency offset.*

I. INTRODUCTION

The main objective of this project to estimate blind Symbol timing and carrier frequency offset acceptable performances of system. In the last years, the interest for filter-bank multicarrier (FBMC) systems is increased, [1] since they provide high spectral containment. Therefore, they have been taken into account for high-data-rate transmissions over both wired and wireless frequency-selective channels. One of the most famous multicarrier modulation techniques is orthogonal frequency division multiplexing (OFDM), other known types of FBMC systems are filtered multi one systems and OFDM based on offset QAM modulation (OQAM).

The FBMC approach complements the FFT with a set of digital filters called polyphase network (PPN) while the OFDM approach inserts the cyclic prefix (CP) after the FFT. Unlike OFDM, OFDM/OQAM systems do not require the presence of a CP in order to combat the effects of frequency selective channels. The absence of the CP implies on the one hand the maximum spectral efficiency and, on the other hand, an increased computational complexity. However, [2] since the sub channel filters are obtained by complex modulation of a single filter; efficient polyphase implementation is often considered.

Fundamental differences between OFDM and OFDM/OQAM systems concern the adoption (in the OFDM/OQAM case) of pulse shaping filters very well localized in time and frequency and memory effects between useful symbols and transmitted signal due to the PPN. OFDM/OQAM systems, as all multicarrier systems, are more sensitive to synchronization errors than single-carrier systems. For this reason, it is very important to derive efficient synchronization schemes. In the last years several studies have been focused on [6] blind and data-aided carrier frequency offset (CFO) and symbol timing (ST) synchronization for OFDM/OQAM systems.

New proposals aim at simplifying the structure of the preamble in order to be able to use it for synchronization and visualization purposes. In synchronization scheme for preamble-based ST and CFO estimation with robust acquisition properties in dispersive channels has been developed [4]. In a new preamble structure has been proposed with useful properties that simplify the use of a one-tap equalizer. The characteristics of the preamble derive from the need to simplify the procedures for channel estimation.

The resulting synchronization algorithms become dependent on the particular preamble, whose utilization is obviously conditioned by the availability of a proper synchronization method. Therefore, [3] a general contribution to the development of synchronization algorithms requires the capability to operate without any specific knowledge about the structure of the preamble. This not only represents a preamble-independent contribution to the synchronization task, which allows a standard definition of the preamble structure unconstrained by the requirements of the synchronization algorithms, but also paves the way to an increase of the spectral efficiency to be achieved by avoiding the preamble. The blind estimation algorithm proposed in is based on the exploitation of the second-order cyclostationarity of the transmitted OFDM/OQAM signal; the convergence of such a method is particularly slow (too many symbol periods have to be processed) so that it is not useful in practice, unless severe signal-to-noise ratios are considered.

It is limited to the case where CFO is present but it is not dedicated to the joint CFO and timing offset estimation, [5] considers the case where both the offsets are jointly estimated by exploiting the cyclostationarity properties. In an algorithm for blind CFO estimation is also proposed according to an approximate (for a large number of subcarriers) maximum-likelihood approach and it is shown its superior performance in comparison with the cyclostationarity-based methods. A maximum likelihood method for blind CFO estimation suited for scenarios of low signal-to-noise ratio is proposed. The weak point of both proposed methods lies in their computational complexity. In this paper, we analyze the conjugate-symmetry property that approximately holds in the beginning of a burst of OFDM/OQAM symbols. Using such an approximate property, a blind method for joint ST and CFO estimation is proposed.

The proposed method is derived with reference to an AWGN channel, it is analyzed by computer simulation with reference to standard multipath channels, and the numerical results show that the proposed method can represent a useful contribution to the blind timing synchronization when the OFDM/OQAM system operates over a multipath channel. The same analysis shows that the proposed method provides a useful contribution to the coarse CFO compensation only for adequate signal-to-noise ratios. [8] Preliminary results about the analysis of the approximate conjugate-symmetry-property in the beginning of a burst of OFDM/OQAM symbols and its exploitation for ST and CFO estimation are reported in organized. The OFDM/OQAM system model is delineated. The conjugate symmetry property (CSP) and the methods to detect it are recalled. It is derived the proposed blind ST estimator exploiting the approximate CSP.

II. ORTHOGONAL FREQUENCY DIVISION MULTIPLEXING

Orthogonal frequency-division multiplexing is a method of encoding digital data on multiple carrier frequencies.[7] OFDM has developed into a popular scheme for wideband digital communication, whether wireless or over copper wires, used in applications such as digital television and audio broadcasting, DSL broadband internet access, wireless networks, and 4G mobile communications. OFDM is essentially identical to coded OFDM (COFDM) and discrete multi-tone modulation, and is a frequency-division multiplexing scheme used as a digital multi-carrier modulation method.

2.1. Basic Architecture of OFDM System

OFDM system block architecture can be divided into 3 main sections, shown in Figure 1, namely the transmitter, the channel and the receiver. The model used in this thesis is tested without the using the Forward Error Correction coding (Denoted in double-line box). The primary advantage of OFDM over single-carrier schemes is its ability to cope with severe channel conditions (for example, attenuation of high frequencies in a long copper wire, narrow band interference and frequency-selective fading due to multipath) without complex equalization filters. Channel equalization is simplified because OFDM may be viewed as using many slowly modulated narrowband signals rather than one rapidly modulated wideband signal.

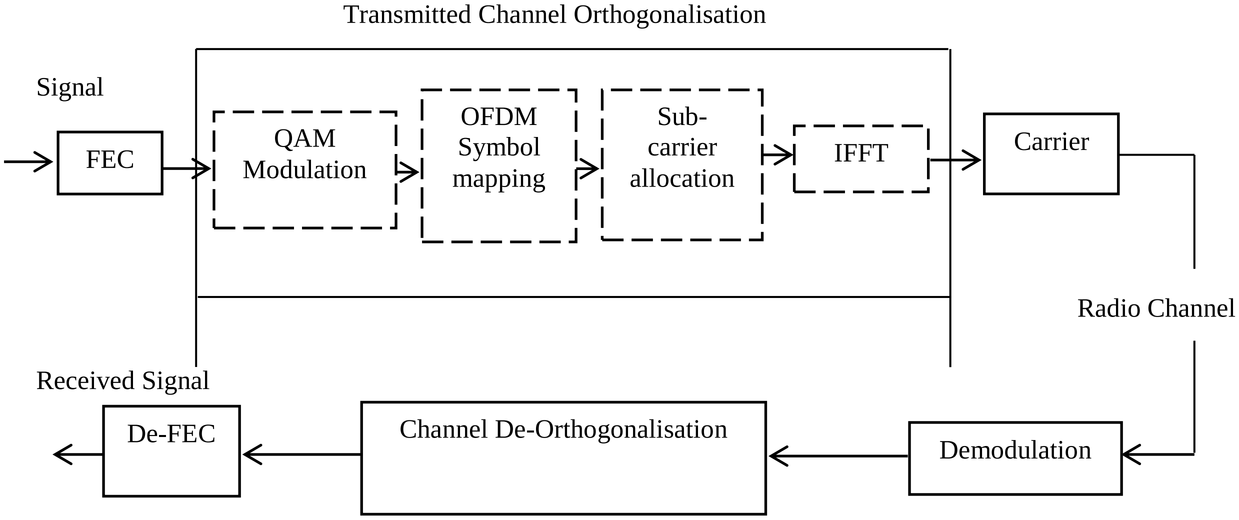

Figure 1. Basic Architecture of OFDM Systems

The low symbol rate makes the use of a guard interval between symbols affordable, making it possible to eliminate intersymbol interference (ISI) and utilize echoes and time-spreading (on analogue TV these are visible as ghosting and blurring, respectively) to achieve a diversity gain, i.e. a signal-to-noise ratio improvement. This mechanism also facilitates the design of single frequency networks (SFNs), where several adjacent transmitters send the same signal simultaneously at the same frequency, as the signals from multiple distant transmitters may be combined constructively, rather than interfering as would typically occur in a traditional single-carrier system.

2.2. Multicarrier System

Multicarrier transmission techniques based on FBMC were developed in the seventies to perform the conversion between Pulse Code Modulation and Frequency Division Multiplexing systems. In the nineties, OFDM was preferred as multicarrier scheme because it was considered simpler in concept, less complex and it had minimum latency. The Figure 2 shows the multicarrier system. OFDM and its variant the OFDMA scheme are the basic communication scheme for the nowadays standards (WLAN, WiMAX, LTE, etc.).

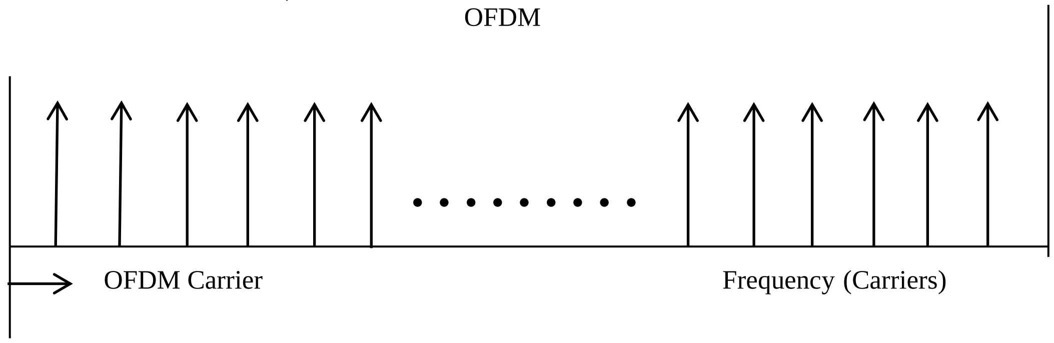

Figure 2. Multicarrier system

2.3. Prototype

Prototype filters are electronic filter designs that are used as a template to produce a modified filter design for a particular application. They are an example of a non-dimensionalised design from which the desired filter can be scaled or transformed. They are most often seen in regards to electronic filters and especially linear analogue passive filters. However, in principle, the method can be applied to any kind of linear filter or signal processing, including mechanical, acoustic and optical filters.
Filters are required to operate at many different frequencies, bandwidth and impedances. The utility of a prototype filter comes from the property that all these other filters can be derived from it by applying a scaling factor to the components of the prototype. The filter design need thus only be carried out once in full, with other filters being obtained by simply applying a scaling factor.

2.4. Synchronization

Synchronization refers to one of two distinct but related concepts: synchronization of processes, and synchronization of data. Process synchronization refers to the idea that multiple processes are to join up or handshake at a certain point, in order to reach an agreement or commit to a certain sequence of action. Data synchronization refers to the idea of keeping multiple copies of a dataset in coherence with one another, or to maintain data integrity. Process synchronization primitives are commonly used to implement data synchronization.

2.5. Signal-to-Noise Ratio

Signal-to-noise ratio is a measure used in science and engineering that compares the level of a desired signal to the level of background noise. It is defined as the ratio of signal power to the noise power, often expressed in decibels. A ratio higher than 1:1 (greater than 0 dB) indicates more signal than noise. While SNR is commonly quoted for electrical signals, it can be applied to any form of signal (such as isotope levels in an ice core or biochemical signaling between cells).
Signal-to-noise ratio is defined as the power ratio between a signal (meaningful information) and the background noise (unwanted signal)

$$SNR=Psignal/Pavg$$

Where *P* is average power.
Both signal and noise power must be measured at the same and equivalent points in a system, and within the same system bandwidth. If the signal and the noise are measured across the same impedance, then the SNR can be obtained by calculating the square of the amplitude ratio:

$$SNR=P_{signal}/P_{noise}=(A_{signal}/A_{noise})^2$$

Where, A is root mean square (RMS) amplitude (for example, RMS voltage). Because many signals have a very wide dynamic range, SNRs are often expressed using the logarithmic decibel scale. In decibels, the SNR is defined as

$$SNR_{dB}=10\log_{10}(P_{signal}/P_{noise}) = P_{signal,dB}-P_{noise,dB}$$

Which may equivalently be written using amplitude ratios as

$$SNR_{dB}=10\log_{10}(A_{signal}/A_{noise})^2= \quad 20\log_{10}(A_{signal}/A_{noise})^2$$

The concepts of signal-to-noise ratio and dynamic range are closely related. Dynamic range measures the ratio between the strongest un-distorted signal on a channel and the minimum discernable signal, which for most purposes is the noise level. SNR measures the ratio between an arbitrary signal level (not necessarily the most powerful signal possible) and noise. Measuring signal-to-noise ratios requires the selection of a representative or reference signal. In audio engineering, the reference signal is usually a sine wave at a standardized nominal or alignment level, such as 1 kHz at +4 dBu1.228 V_{RMS}).SNR is usually taken to indicate an average signal-to-noise ratio, as it is possible that (near) instantaneous signal-to-noise ratios will be considerably different. The concept can be understood as normalizing the noise level to 1 (0 dB).

2.6. Root Mean Square Error (RMSE)

The root-mean-square deviation (RMSD) or root-mean-square error (RMSE) is a frequently used measure of the differences between values predicted by a model or an estimator and values actually observed. These individual differences are called residuals when the calculations are performed over the data sample that was used for estimation, and are called prediction errors when computed out-of-sample. The RMSD serves to aggregate the magnitudes of the errors in predictions for various times into a single measure of predictive power. RMSD is a good measure of accuracy, but only to compare forecasting errors of different models for a particular variable and not between variables, as it is scale-dependent.

III. EXISTING METHOD

Additive white Gaussian noise (AWGN) shown in Figure 3 is a channel model in which the only impairment to communication is a linear addition of wideband or white noise with a constant spectral density(expressed as watts per hertz of bandwidth) and a Gaussian distribution of amplitude. The model does not account for fading, frequency selectivity, interference, nonlinearity or dispersion. It produces simple and tractable mathematical models which are useful for gaining insight into the underlying behaviour of a system before these other phenomena are considered. Wideband Gaussian noise comes from many natural sources, such as the thermal vibrations of atoms in conductors (referred to as thermal noise or Johnson-Nyquist noise), shot noise, black body radiation from the earth and other warm objects, and from celestial sources.

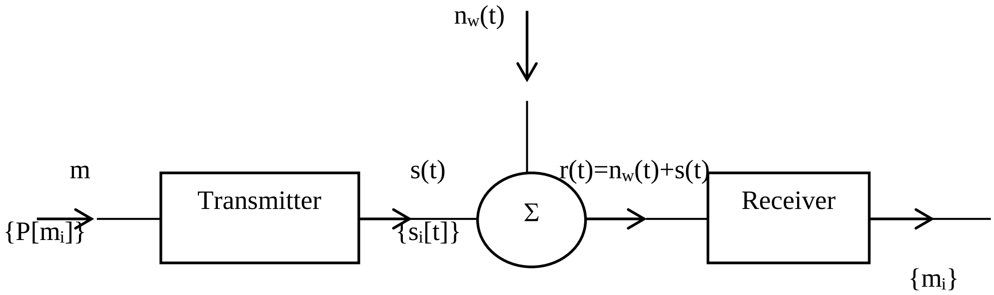

Figure 3. Additive White Gaussian Noise

The AWGN channel is a good model for many satellite and deep space communication links. It is not a good model for most terrestrial links because of multipath, terrain blocking, interference, etc. However, for terrestrial path modeling, AWGN is commonly used to simulate background noise of the channel under study, in addition to multipath, terrain blocking, interference, ground clutter and self interference that modern radio systems encounter in terrestrial operation.

3.1. Filter Bank Multicarrier (FBMC)

Filter bank Multicarrier in the system, the efficiency of present OFDM based solutions can be increased while conserving high degree of compatibility with the existing equipment. Filter bank Multicarrier Technology avoids spectral waste and provided better frequency localization by introducing an efficient pulse shaping in the modulation scheme, avoiding distortion from non-synchronous signals in adjacent bands. In order to enable the use of accurately nonrectangular pulse-shaping, the different subcarriers need to be modulated using staggered offset QAM modulation. The application of this modulation presents an additional advantage. The filter bank multicarrier is less sensitivity to frequency offsets. Filter bank Multicarrier Technology increases the data rate since it does not use any cyclic prefix to combat channel effects. Filter bank Multicarrier Technology is a strong candidate envisioned for high speed PLC due to its high spectral efficiency in terms of bps/Hz, frequency properties, which fits the stringent frequency masks imposed for PLC links and high level of compatibility with the OFDM based physical layer defined in the standard.

3.2. ITU Channel

The International Telecommunication Union (ITU) is an agency of the United Nations (UN) whose purpose is to coordinate telecommunication operations and services throughout the world. Originally founded in 1865, as the International Telegraph Union, the ITU is the oldest existing international organization. ITU headquarters are in Geneva, Switzerland.
The ITU consists of three sectors:

- Radio communication (ITU-R) -- ensures optimal, fair and rational use of the radio frequency (RF) spectrum.
- Telecommunication Standardization (ITU-T) - formulates recommendations for standardizing telecommunication operations worldwide.
- Telecommunication Development (ITU-D) -- assists countries in developing and maintaining internal communication operations.

The ITU sets and publishes regulations and standards relevant to electronic communication and broadcasting technologies of all kinds including radio, television, satellite, telephone and the Internet. The organization conducts working parties, study groups and meetings to address current and future issues and to resolve disputes. The ITU organizes and holds an exhibition and forum known as the Global TELECOM every four years. Another important aspect of the ITU's mandate is helping emerging countries to establish and develop telecommunication systems of their own. Although the recommendations of the ITU are non-binding, most countries adhere to them in the interest of maintaining an effective international electronic communication environment.

IV. PROPOSED METHOD

In a typical wireless communication system, the signal to be transmitted is up converted to a carrier frequency prior to transmission. The receiver is expected to tune to the same carrier frequency for down converting the signal to baseband, prior to demodulation. The block diagram of carrier frequency offset is shown in Figure 4.

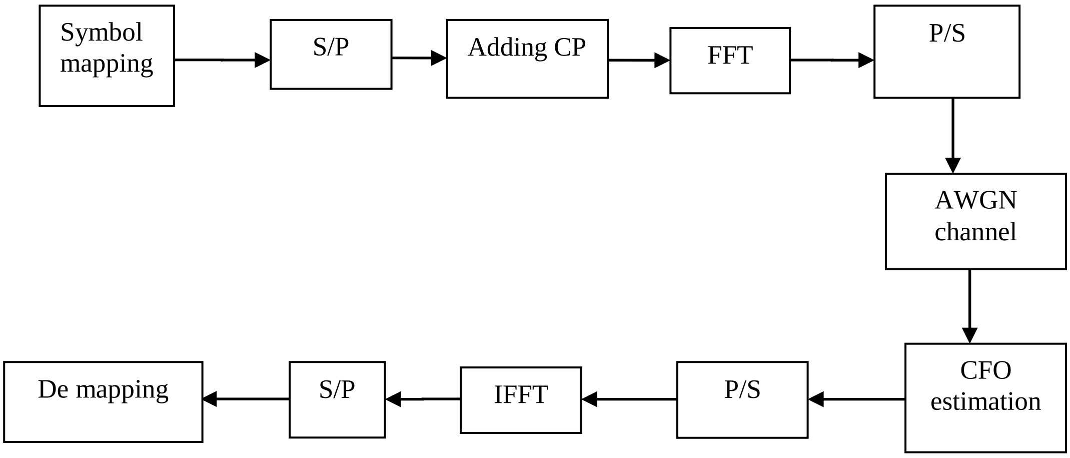

Figure 4. Block Diagram of Carrier Frequency Offset

4.1. Fast Fourier Transform

A fast Fourier transform (FFT) is an algorithm to compute the discrete Fourier transform (DFT) and it's inverse. A Fourier transform converts time (or space) to frequency and vice versa; an FFT rapidly computes such transformations. As a result, fast Fourier transforms are widely used for many applications in engineering, science, and mathematics. Fast Fourier transforms have been described as "the most important numerical algorithm of our lifetime.

The DFT is obtained by decomposing a sequence of values into components of different frequencies. This operation is useful in many fields (see discrete Fourier transform for properties and applications of the transform) but computing it directly from the definition is often too slow to be practical. An FFT is a way to compute the same result more quickly: computing the DFT of N points in the naive way, using the definition, takes $O(N^2)$ arithmetical operations, while a FFT can compute the same DFT in only $O(N \log N)$ operations. The difference in speed can be enormous, especially for long data sets where N may be in the thousands or millions. In practice, the computation time can be reduced by several orders of magnitude in such cases, and the improvement is roughly proportional to $N / \log(N)$. This huge improvement made the calculation of the DFT practical; FFTs are of great importance to a wide variety of applications, from digital signal processing and solving partial differential equations to algorithms for quick multiplication of large integers.

The best-known FFT algorithms depend upon the factorization of N, but there are FFTs with $O(N \log N)$ complexity for all N, even for prime N. Many FFT algorithms only depend on the fact that $e^{-2\pi i/N}$ is an N-th primitive root of unity, and thus can be applied to analogous transforms over any finite field, such as number-theoretic transforms. Since the inverse DFT is the same as the DFT, but with the opposite sign in the exponent and a $1/N$ factor, any FFT algorithm can easily be adapted for it.

4.2. Cyclic Prefix

Cyclic prefix is often used in conjunction with modulation in order to retain sinusoids properties in multipath channels. It is well known that sinusoidal signals are eigen functions of linear and time-invariant systems. Therefore, if the channel is assumed to be linear and time-invariant, then a sinusoid of infinite duration would be an eigen function. However, in practice, this cannot be achieved, as real signals are always time-limited. So, to mimic the infinite behavior, prefixing the end of the symbol to

the beginning makes the linear convolution of the channel appear as though it were circular convolution and thus, preserve this property in the part of the symbol after the cyclic prefix.
Cyclic Prefixes are used in OFDM in order to combat multipath by making channel estimation easy. As an example, consider an OFDM system which has N subcarriers. The message symbol can be written as:

$$d = [d_0, d_1, \dots d_{N-1}]$$

The OFDM symbol is constructed by taking the inverse discrete Fourier transform (IDFT) of the message symbol, followed by a cyclic prefixing. Let the symbol obtained by the IDFT be denoted by

$$X' = [x[0], x[1], \dots x[N-1]]^T$$

Prefixing it with a cyclic prefix of length, L-1 the OFDM symbol obtained

$$X = [x[]N-L+1], .x[N-2], x[N-1], x[0], x[1], \dots x[N-1]]^T$$

Assume that the channel is represented using

$$h = [h_0, h_1, \dots . h_{L-1}]^T$$

So, taking the Discrete Fourier Transform, we get

$$y[k] = H[k] \,.\, X[k]$$

Where X[k] is the discrete Fourier transform of X. Thus, a multipath channel is converted into scalar parallel sub-channels in frequency domain, thereby simplifying the receiver design considerably. The task of channel estimation is simplified, as we just need to estimate the scalar coefficients H[k] for each sub-channel and once the values of are estimated, for the duration in which the channel does not vary significantly, merely multiplying the received demodulated symbols by the inverse of yields the estimates of and hence, the estimate of actual symbols.

4.3. Inverse Fast Fourier Transform

The inverse Fast Fourier Transform is a common procedure to solve a convolution equation provided the transfer function has no zeros on the unit circle. In our paper we generalize this method to the case of a singular convolution equation and prove that if the transfer function is a trigonometric polynomial with simple zeros on the unit circle, then this method can be extended.

4.4. Symbol Timing

The symbol timing (also known as baud or modulation rate) is the number of symbol changes (waveform changes or signaling events) made to the transmission medium per second using a digitally modulated signal or a line code. The Symbol rate is measured in baud rate or symbols/second. In the case of a line code, the symbol rate is the pulse rate in pulses/second. Each symbol can represent or convey one or several bits of data. The symbol rate is related to, but should not be confused with, the gross bit rate expressed in bit/second.
A symbol can be described as either a pulse (in digital baseband transmission) or a "tone" (in pass band transmission using modems) representing an integer number of bits. A theoretical definition of a symbol is a waveform, a state or a significant condition of the communication channel that persists for a fixed period of time. A sending device places symbols on the channel at a fixed and known symbol rate, and the receiving device has the job of detecting the sequence of symbols in order to reconstruct the transmitted data. There may be a direct correspondence between a symbol and a small unit of data (for example, each a symbol may encode one or several binary digits or 'bits') or the data may be represented by the transitions between symbols or even by a sequence of many symbols. The symbol duration time, also known as unit interval, can be directly measured as the time between transitions by looking into an eye diagram of an oscilloscope. The symbol duration time T_s can be calculated as:

$$T_s = 1/f_s$$

Where, fs is the symbol rate.

V. OUTPUT OF PROPOSED METHOD

The output of OQAM modulation is shown in Figure 5.

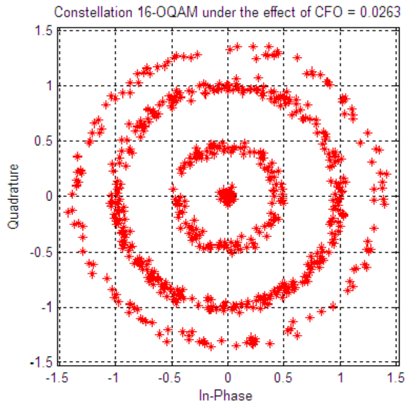

Figure 5. Output of OQAM

The output for estimating CFO in OQAM modulation in AWGN channel is shown in Figure 6.

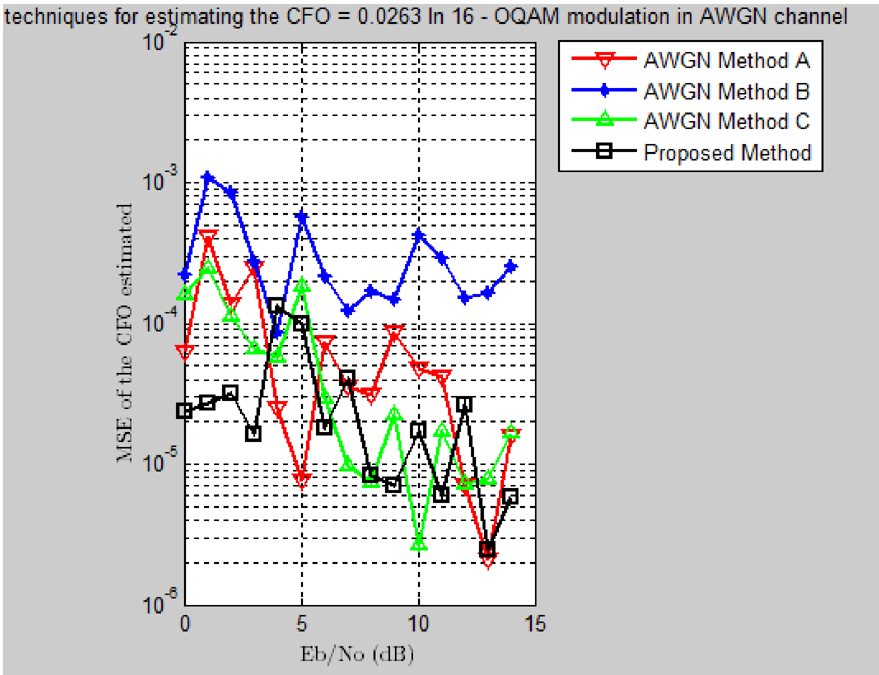

Figure 6. Output of estimating CFO

VI. Conclusions

The problem of blind synchronization for OFDM/OQAM systems has been considered. Specifically, a new method for blind ST and CFO synchronization has been proposed by exploiting the approximate CSP of the beginning of a burst of OFDM/OQAM symbols due to the presence of the time offset. The results of the performance analysis with reference to the considered OFDM/OQAM system show that the proposed blind ST and CFO estimators, complemented by a simpler coarse ST estimator, achieve acceptable performance for realistic values of *Eb/N*0.

References

[1]. Bellanger. M, "Efficiency of filter bank multicarrier techniques in burst radio transmission," *in Proc. IEEE Global Commun..conf., pp.1-4*

[2]. Cherubini. G, Eleftheriou. E, Oker. S, and Cioffi. J, (2000) "Filter bank modulation techniques for very high speed digital subscriber lines,"*IEEECommun. Mag…vol.38, pp. 98-104.*

[3]. Ciblat. P and Serpedin. E, (2004) "A fine blind frequency offset estimator for OFDM/OQAM systems," *IEEE Trans. Signal Process..vol.52, pp.291-296.*

[4]. DavideMattera and Mario Tanda“Blind Symbol Timing and CFO Estimation for OFDM/OQAM Systems”*IEEE transactions on wireless communications,.*
[5]. Fusco. T and Tanda. M, (2007) “Blind frequency-offset estimation for OFDM/OQAM systems,” *IEEE Trans. Signal Process..vol.55, pp.1828-1838.*
[6]. Fusco. T, Petrella. A, and Tanda. M, (2009) “Data-aided symbol timing and CFO synchronization for filter-bank multicarrier systems,”*IEEETrans.WirelessCommun..vol.8, pp.2705-2715.*
[7]. Le Floch. B, Alard. M, and Berrou. C, “Coded orthogonal frequency division multiplex,” *Proc. IEEEvol. 83, pp. 982–996.*
[8]. Mattera. D and Tanda.M, “A new method for blind synchronization for OFDM/OQAM systems,” *in Proc.,International Symp.Image Signal Process,. Analysis.*