

Optimal Parameters of the IQ Imbalance Correction Algorithm Based on Adaptive Filter

Miljan Petrović, Miljana Milić, Srđan Milenković, *University of Niš*

Abstract—In phase (I) and quadrature (Q) complex signal representation ($I+jQ$) of the modulated signal is widely used in RF systems. Ideally, I and Q signals have equal amplitude while Q is 90° phase shifted relative to I in the case of SSB (single-sideband) modulations. Having quadrature signaling provides many advantages such as higher RF spectrum efficiency (more bits/Hz), lower data converters (ADCs/DACs) sampling rate for the same data throughput, computational power of the base band modem is relaxed, all making low power transceiver implementations possible. However, any mismatch of gain or phase between I and Q (IQ imbalance) will degrade transceiver performance. Error vector magnitude (EVM), bit error rate (BER) and RX sensitivity in homodyne radios are among the most critical parameters affected. There are multiple sources of IQ imbalance. RF mixers can have different gain for I and Q paths, and the gain can also be frequency dependent. PLL which is responsible to generate quadrature LO produces nonequal I and Q signals in terms of phase shift. Even base band modules such as low pass filters or gain stages can contribute. The most difficult to estimate during the system design phase is in fact IQ cross talk happening both on the PCB and inside the RF IC. This paper provides insight into an adaptive filtering method for the correction of the I/Q imbalance in the receive chain. The image rejection is evaluated using 64-QAM and 16-QAM test signals for different adaptive filter designs. The efficiency of the algorithm is inspected in the parameter space consisting of the adaptation step-size and the order of the adaptive filter. Simulation results show that zero order filter and reasonably low value of step-size provide optimum filtering for the above mentioned test cases. Constellation diagrams of the simulated received and adaptively filtered signals illustrate the importance of the adaptive filter key parameters setting.

Index Terms—Adaptive filtering; I/Q imbalance; RF communications; QAM; Quadrature signal processing.

I. INTRODUCTION

The computational power of modern processors facilitates the analysis of vast volumes of information in real-time. Real world signals are continuous in amplitude and time, and need to be represented in digital form by discretization in time and quantization in amplitude so that digital computer can process them. Digital signal processing can be found in many different

applications. DSP algorithms are predictable and repeatable for the same given inputs stream. This further gives the advantage of easy simulation and short design time [1].

In order to improve the DSP power, complex number mathematics can be applied, offering methods that cannot be realized with real number mathematics. Complex DSP is more abstract and theoretical than the real DSP. Nevertheless, complex DSP is more powerful and comprehensive. Methods and transformations over the complex numbers, such as complex modulations, filtering, mixing, z-transform, speech analysis and synthesis, adaptive complex processing, complex Fourier transforms etc., are the essential parts of the complex DSP. Many wireless high-speed telecommunication standards (Wi-Fi, 3G, 4G and soon coming 5G, for example) require complex DSP methods. Complex signals are very common in telecommunications and the corresponding complex methods became inevitable [2].

I/Q (In phase / Quadrature) signals are often used in RF applications. They form the basis of the complex RF signal modulation, demodulation, and complex signal analysis. The In-phase signal is referred to as I, and the signal Q has all frequency components shifted by 90° in the case of SSB (single-sideband) modulations [3]. The amplitude and the phase of the sum of the quadrature signals are functions of the values of I and Q. Therefore, one can create modulated RF signals by varying I and Q- signals over time. In other words, using this technique, one does not need to directly vary the phase of an RF carrier (LO). Complex quadrature signal processing is a crucial concept in radar systems, coherent processing, antenna beamforming applications, mobile communications, etc [4].

Some advantages of I/Q signal representations are as follows: for the same signal bandwidth, one can use half the sampling rate of the standard real-signal sampling, which further provides low power implementation; calculation of FFT (Fast Fourier Transform) is more computationally efficient; instantaneous amplitude and phase can be accessed via I and Q signals.

However, ideal phase shift and consequently, ideal I/Q signal representation is more difficult to achieve in the real radio system due to mismatches in the analog and RF receiver blocks. The purpose of this paper is to examine a method for I/Q imbalance correction based on adaptive digital filtering of the complex symbol sequence. First, I/Q structure of the receiver is described. Further, I/Q imbalance problem is presented and some developed solutions are reviewed. Finally, the adaptive correction filter is described and

Miljan Petrović – Faculty of Electronic Engineering, University of Niš, Aleksandra Medvedeva 14, 18000 Niš, Serbia (e-mail: miljan.petrovic@elfak.ni.ac.rs).

Miljana Milić – Faculty of Electronic Engineering, University of Niš, Aleksandra Medvedeva 14, 18000 Niš, Serbia (e-mail: miljana.milic@elfak.ni.ac.rs).

Srđan Milenković – Faculty of Electronic Engineering, University of Niš, Aleksandra Medvedeva 14, 18000 Niš, Serbia (e-mail: srdjan@milenkovici.net).

designed for 64-QAM and 16-QAM signals. The aim of the paper is to find patterns in the procedure of estimating the optimal step-size and filter order. Conclusions about the effects of these parameters on the I/Q imbalance correction algorithm are given at the end.

II. I/Q RECEIVER DESIGN

A. Receiver Structure

The structure of an I/Q RF receiver is given in Fig. 1. First, the signal coming from antenna point and amplified by low noise amplifier (LNA) is mixed with the quadrature carrier signal generated by the PLL. This shifts the signal from RF back to baseband frequency. The two outputs are passed through the identical anti alias/channel select low pass filters, and further sampled and quantized by A/D converters. Finally, we get the in phase and quadrature signals ready for further processing. It should be noted that for a correct functioning of the adaptive algorithm (that will be described in the next section), a symbol detector in the digital domain should be implemented between the receiver structure of Fig. 1 and the adaptive filter. Often, as shown in Fig. 1, programmable gain stages are placed between the blocks in order to fulfill the dynamic range specifications of the consequent block.

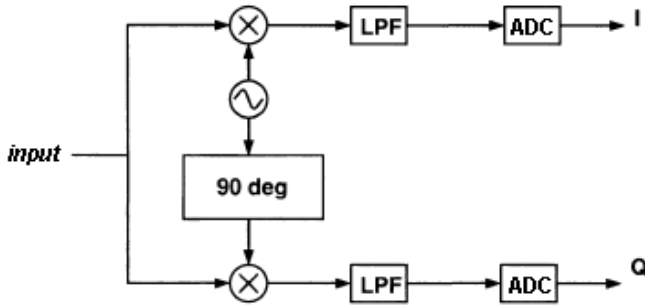


Fig. 1. Schematic of the RF receiver structure.

B. I/Q Imbalance Problem

Ideal I and Q signals are 90° phase shifted and have identical amplitude in the case of SSB modulations, which means that the spectrum of the signal $I+jQ$ only consists of positive frequency components (in the baseband). However, mismatches of electronic components often lead to problem known as the I/Q imbalance. In general, it is difficult to achieve symmetry in analog circuitry. Therefore, carrier signals are not exactly 90° shifted, frequency responses of the low pass filters are slightly different, layout design introduces different path delays for I and Q signals, capacitors mismatches in filters and A/D converters can induce interference, etc [5]. It is known that wideband I and Q signals have frequency dependent imbalance, whereas narrowband signals have frequency independent imbalance.

I/Q imbalance is illustrated in Fig. 2 assuming low IF receiver. The wanted signal spectrum is centered around the frequency f_0 . Ideally, there should be no signal components in the range $f < 0$. However, due to the I/Q imbalance, a signal

image is present. In zero IF receivers, $f_0 = DC$, which means that IQ imbalance image is inside the wanted signal frequency range and as such cannot be filtered at all. Hence a sophisticated algorithm for IQ imbalance cancellation must be implemented.

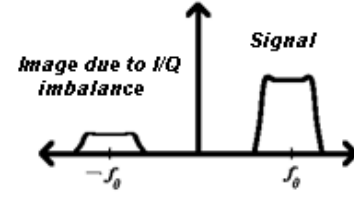


Fig. 2. Illustration of the spectrum of an imbalanced I/Q signal.

C. Solutions

Since I/Q imbalance is a very common problem in telecommunications, many techniques for reducing its effects have been derived [6]. Here is a short review of the most popular ones.

First, there is the off-line adjustment presented in [7]. The method uses Gram-Schmidt procedure. Although it provides good results, it is constrained by the facts that it is not a real time algorithm and that it can only be applied to signals with frequency independent mismatches, i.e. narrow band I and Q signals.

Another solution is to develop the analog RF receiver with only one branch (only I output), and then generate the quadrature signal in the digital domain [8]. This method is valid for SSB modulations, and it incorporates a digital Hilbert transformer. Since a high-quality digital filter approximation of the Hilbert transformer usually needs a high filter order, this method leads to disadvantages such as high delay and power consumption, and low processing speed [9].

The most popular I/Q imbalance correction method is the application of adaptive filters. A wide range of problems is covered by adaptive filters, since they can be used in several classes of problems: identification and inverse modeling, signal prediction, interference and noise cancellation [1]. In the case of the I/Q imbalance, noise cancellation structure is used. However, certain modifications in the filter structure and/or in the adaptive algorithm are needed to address the specific problem. For example, in [9] a complex adaptive filter is derived filtering both the reference input and the primary input. Standard structure only filters the reference input. An adaptive filter presented in [10], provides excellent results in complex baseband I/Q imbalance correction.

III. ADAPTIVE I/Q IMBALANCE CANCELLATION

A. Adaptive Algorithm

The adaptive filter evaluated in this paper is described in more details in [10]. The filter's primary input is the symbol complex sequence $I+jQ$ derived from I and Q channels after signal detection. Each sample represents one received symbol.

The schematic of the filter is given in Fig. 3.

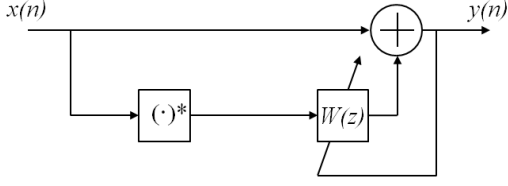


Fig. 3. Schematic of the adaptive filter for compensating I/Q imbalance.

The reference input of the filter is the complex conjugate of the primary input signal. The reference signal is filtered through adaptive filter with coefficients w , and then added to the primary input, thus forming the filter output. This can be mathematically presented together with the coefficients update expression as:

$$\begin{aligned} y(n) &= x(n) + \overline{x(n)}^* w_m(n) \\ w_{m+1} &= w_m - \mu \cdot y^2 \end{aligned} \quad (1)$$

where x denotes the primary input, y the filter output, w_m is the set of complex coefficients in the m^{th} iteration, and n is the time index. This algorithm ensures that the mathematical expectation $E[y^2]$ converges to zero [10]. Since the input consists of complex symbols, the algorithm tends to rearrange them in the constellation diagram so that they are symmetrically ordered around the point $0+j0$. Implicitly, this moves the symbols towards the reference points.

B. Setting Step-Size and Filter Order

After selecting the right adaptive algorithm, the next step in I/Q imbalance correction is to define the adaptive filter's parameters. Two parameters are necessary to set: adaptation step-size μ , and the filter order N . Effects of these parameters on the filter performance greatly depend on the specific application. Here, the parameter space is discussed in the context of I/Q imbalance correction. The errors of received and filtered signals are evaluated in many points in the parameter space for two types of complex modulations: 64-QAM and 16-QAM.

1) 64-QAM Case

Each sample of the received signal represents a complex symbol, which is, due to mismatches, at some distance (Euclidean) from the reference symbol point in the constellation diagram. When the adaptive algorithm converges, one should expect these distances to be lower. The mean value of all distances in the 3500 samples long signal (which are randomly generated symbols) is chosen as the error function. This function is evaluated for the filter orders from 0 to 4, and step-sizes in the range from 10^{-6} to 10^{-3} . The values are given in Fig. 4.

IQ imbalance in the received signal is simulated as changes in amplitude and phase of generated symbols. Amplitude error is set to 2dB and phase error to 10° .

It can be seen that filters with lower order achieve smaller

errors. This is why the optimal filter order is obviously equal to 0. Higher orders lead to overfitting the signal model, which is manifested as the presence of the noise at the output because the algorithm considers it as a signal component. The error function due to step-size for a constant filter order has a global minimum. Too low value of the step-size does not significantly change the filter coefficients, which makes the convergence so slow that the filter can be considered to diverge for practical purposes. Hence, the filter output is approximately the same as the filter input. On the other hand, too large value of the step-size makes the algorithm diverge, and noise is being amplified instead of cancelled. This is manifested as diverging from the minimum of the cost function of the adaptive algorithm.

Fig. 4 suggests that the error function (with respect to the step-size) of the higher order filter, has no minimum. However, this is not the truth, since the minimum is just located in a very low order of the magnitude range of values and cannot be seen on the plot. In this case, very low value of step-size, even if the rendered error is the global minimum of the parameter space, is not feasible for a finite-length digital implementation of the filter. This is why the possibly suboptimal solution is chosen as the minimum of the error function in the case of filter order 0.

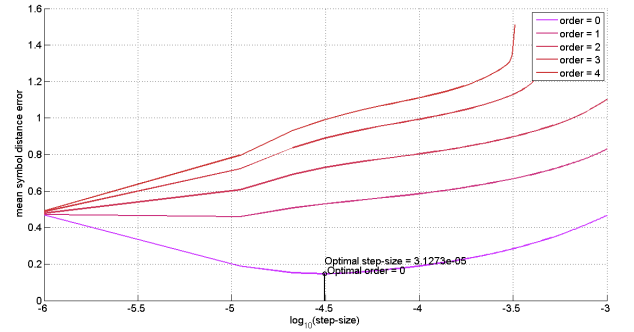


Fig. 4. Log-linear plot of mean symbol distance error dependence on the step-size for several filter orders, in the case of 64-QAM.

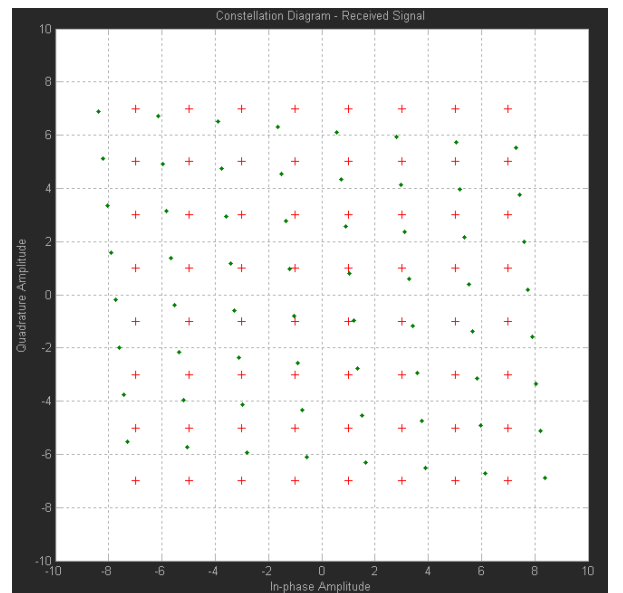


Fig. 5. Constellation diagram of the received 64-QAM signal.

Figs. 5. and 7. show constellation diagrams of the received and filtered signal in the case of the chosen (sub)optimal parameters. Step-size was set to $3.1273 \cdot 10^{-5}$, and filter order to 0. The comparison of diagrams suggests more dispersion of the detected symbols in the received signal, whereas symbols are more accurately concentrated around reference (target) symbols in the case of filtered signal. Further, one can notice that IQ imbalance manifests in a way that it makes rectangular constellation diagram resembles to a rhomboid. Finally, Fig. 6 illustrates the mean distances (mean errors) for each symbol separately (with integer values from 0 to 63), before and after filtering. Black lines denote errors of the received signal, and blue lines represent errors of the filtered signal. Significant decrease of these errors due to adaptive filtering is noticed.

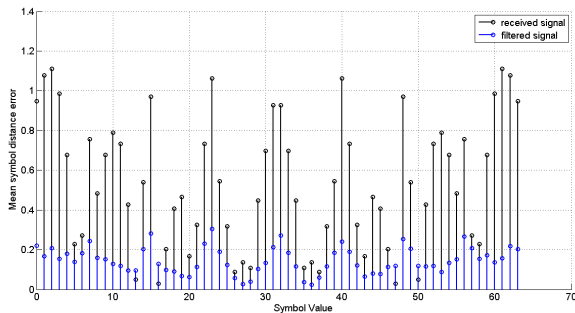


Fig. 6. Mean distance errors for each symbol in 64-QAM signal before and after adaptive filtering.

2) 16-QAM

Similarly to the presented analysis of the 64-QAM signal, 16-QAM modulation technique is used to evaluate the IQ imbalance correction algorithm. In Fig. 8, error function in the parameter space is displayed. The same remarks can be made as in the case of 64-QAM. Also, it can be concluded that the optimal step-size is greater in the case of 16-QAM. In this case, optimal parameters are step-size $1.8264 \cdot 10^{-4}$, and filter order 0.

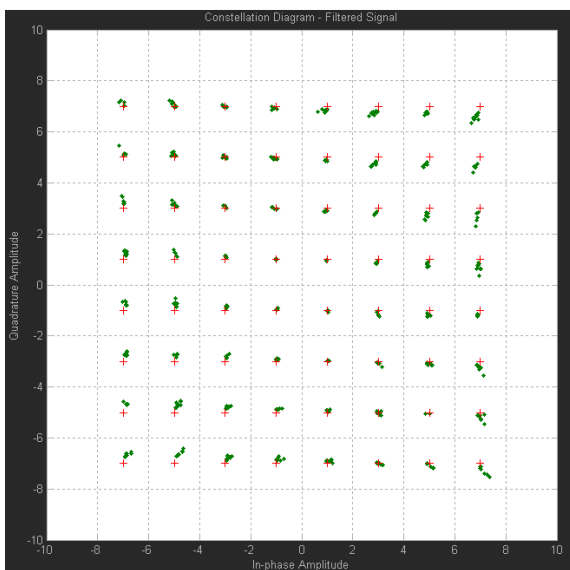


Fig. 7. Constellation diagram of the 64-QAM signal after adaptive filtering with optimal parameters.

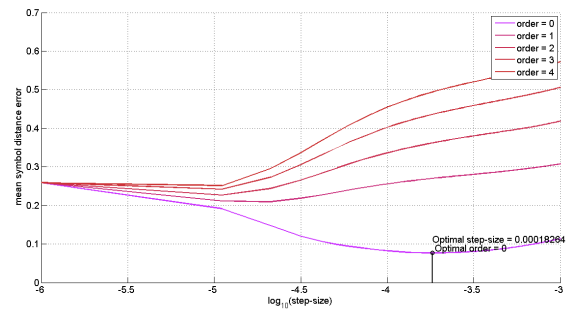


Fig. 8. Log-linear plot of mean symbol distance error dependence on the step-size for several filter orders, in the case of 16-QAM.

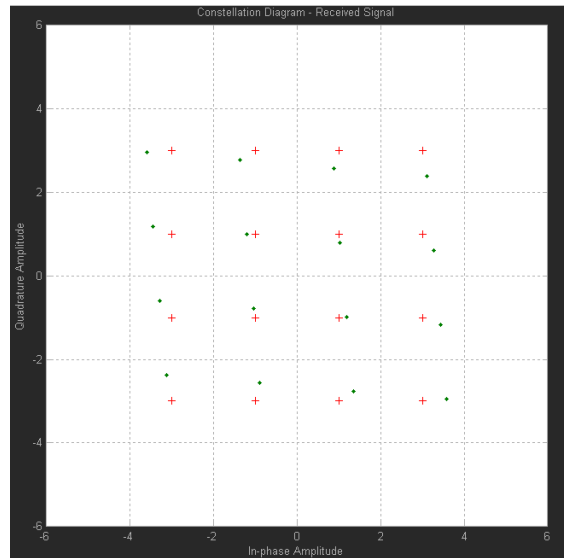


Fig. 9. Constellation diagram of the received 16-QAM signal.

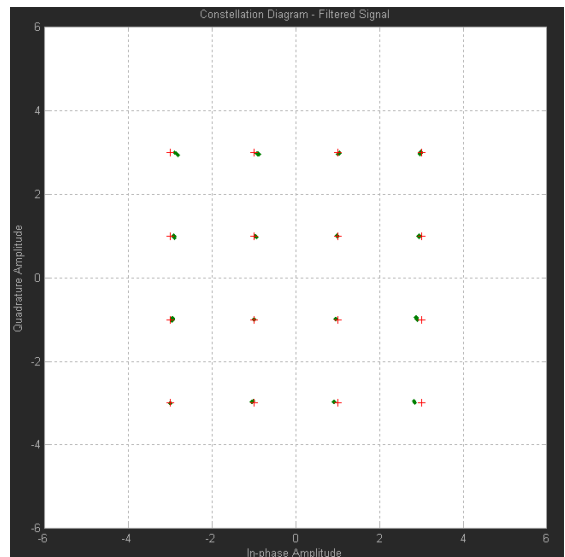


Fig. 10. Constellation diagram of the 16-QAM signal after adaptive filtering with optimal parameters.

This suggests an order of magnitude higher values of the optimal step-size in the case of 16-QAM, which can be explained by the fact that less number of symbols in the modulation technique means less variability in the received

signal, which further means that the adaptive algorithm does not need too low step-size in order to converge. Figs. 9. and 10. show constellation diagrams of the received and filtered 16-QAM signals. Further, Fig. 11 displays mean errors for each symbol separately (with integer values from 0 to 15), before and after filtering. Black lines denote errors of the received signal, and blue lines represent errors of the filtered signal.

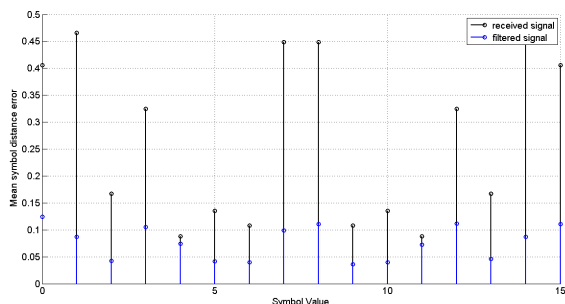


Fig. 11. Mean distance errors for each symbol in 16-QAM signal before and after adaptive filtering.

IV. CONCLUSION

This paper has analyzed the design of I/Q imbalance correction adaptive filter. Step-size and filter order were assessed in the case of 64-QAM and 16-QAM complex signals. It was shown that the optimal order is the lowest one, i.e. zero order. This means that there is only one complex coefficient (two real ones – its real and imaginary part), which makes the algorithm converge faster. In the case of 64-QAM, the step-size is of order of magnitude 10^{-5} , and in the case of 16-QAM, 10^{-4} . This fact is explained by the higher number of symbols rendering the algorithm harder to converge, thus moving the optimal step-size to a lower value.

The presented analysis is valid for the QAM modulation technique. Since adaptive algorithms parameters greatly depend on the specific application and the nature of the input signals, other types of modulations should be explored. Also, in order to completely evaluate the I/Q imbalance correction

method, the adaptive filter should be explored together with the appropriate symbol detection system. Different methods of transforming I+jQ signal into a signal, in which each sample represents one symbol, need to be analyzed.

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