ON ADAPTIVE CANCELLATION OF IQ MISMATCH IN OFDM RECEIVERS

S. FOULADIFARD, H. SHAFIEE
Dept. of Electrical and Computer Engineering, University of Tehran, Tehran, Iran, hshafiee@ut.ac.ir

Abstract – Design of wireless transceivers are driven by market demands for low-power and low cost terminals. Direct conversion receivers have thus been the focus of active research in recent years, since they offer great potentials for compact and low power solutions. However, most direct conversion designs suffer from gain and phase imbalances between the in-phase and quadrature paths. In this paper, two adaptive algorithms for compensation of IQ mismatch effects in OFDM systems are evaluated and compared. One technique uses the samples in the time-domain while the other uses an adaptive two-tap equalizer for each sub-channel. The results of computer simulations show that using the time-domain approach, IQ imbalance can be almost completely compensated.

I. INTRODUCTION

Architectures for radio receivers are driven by market demands for low-cost and low power solutions for wireless communications. Toward that end, the implementation of direct conversion receivers, which in contrast to classical heterodyne architectures convert the radio frequency (RF) signal directly to baseband, has been the subject of active research in recent years [1-2].

While efficient in terms of power consumption, the practical design of direct conversion mixers introduces certain non-ideal effects in the transmitter (TX) and receiver (RX). For example, spurs and DC offsets (due to local oscillator leakage) are more significant in such designs. In addition, the so-called IQ mismatch effect is more pronounced in direct conversion architectures compared to heterodyne and other architectures, since in the latter implementations, better component matching can generally be attained. IQ mismatch refers to phase and gain imbalance between in-phase (I) and quadrature (Q) paths [1-2]. Specifically, phase mismatch occurs when the phase difference between the local oscillator signals for I and Q channels is not exactly 90 degrees. Gain imbalance refers to a gain mismatch in the path of the I and Q signals.

Such effects cause a distortion of the received signal as well as a rotation of the symbols in the constellation space and degrade the bit-error-rate performance of the system considerably. It is, therefore, imperative that compensation mechanisms are designed to counter the effects of IQ mismatch. The amount of gain and phase mismatches can change with frequency as the local oscillator frequency changes as well as with time and ambient temperature. Therefore, any compensation algorithm needs to work in an adaptive way, or be re-tuned periodically in a calibration mode.

In [3], we proposed an adaptation scheme for cancellation of IQ mismatch for an orthogonal frequency division multiplexing (OFDM) system. Due to its spectral efficiency and its effectiveness in frequency selective fading channels, OFDM has been considered as the preferred signaling technique for a variety of communication systems, especially in wireless applications. For example, IEEE standards 802.11a and 802.11g for wireless local area networking include OFDM as a mandatory signaling technique [4].

The adaptive IQ canceller in [3] uses the baseband time-domain samples, i.e., complex samples prior to the discrete Fourier transform operation in the OFDM receiver. An alternate adaptive structure is proposed in [5] which uses two-tap equalizers that operate on the frequency-domain samples, i.e., those in each sub-channel after the DFT block. In this paper, we evaluate and compare the performance of these two adaptive techniques for cancellation of IQ imbalance.

This paper is organized as follows: in the next section, the effect of IQ imbalance in an OFDM system is discussed. Next, we describe the two adaptive compensation schemes. In section V, we present the simulation results of applying the two proposed techniques and compare their merits. Finally, conclusions are summarized in Section VI.

II. IQ MISMATCH IN OFDM TRANSCEIVERS

In OFDM signaling, symbols are modulated onto one of N orthogonal sub-carriers, which are equally separated by $\Delta f = 1/T_s$ Hz, where T_s is the duration of sub-carrier waveforms. By adding the sub-carrier modulated signals together and sampling the resulting waveform at instants $t = nT_s / N$, the OFDM-modulated sequence is obtained which is given by

$$x[n] == \frac{1}{N} \sum_{k=-K}^{K} X[k] e^{+j2\pi k \frac{n}{N}} \tag{2.1}$$

where $0 \le n \le N$-1. In addition, it is considered that only sub-carriers in the range [-K, K] where $N \ge (2K+1)$, contain non-zero symbols. Equation (2.1) is equivalent to an inverse discrete Fourier transform of the modulated data symbols $X[k]$ and therefore, could be implemented using inverse fast Fourier transform (IFFT). A complete OFDM symbol is formed by adding N_g guard samples from the end of each block to the beginning. The real and imaginary components of the IFFT output are converted to continuous-time waveforms using digital-to-analog converters and then low-pass filtered. The signal is then up-converted to the desired carrier frequency, f_c , amplified and transmitted.

At the receiver, quadrature demodulation of the received RF waveform followed by low–pass filtering yields the I and Q signals. The continuous-time I and Q signals are then sampled using a pair of analog-to-digital converters (ADC). With the length of the guard interval chosen to be longer than the longest delay spread expected, the linear convolution of the transmit sequence and the channel response becomes equivalent to circular convolution. In the frequency domain, we thus have

$$Y[k] = X[k]\,H[k] + N[k], \quad k = -K,...,K \tag{2.2}$$

where $Y[k]$ denotes the received signal at sub-carrier k, $H[k]$ is the corresponding channel response and $N[k]$ shows the effect of additive noise. To counter the effect of channel response, a one-tap equalizer operates on symbols in each sub-channel such that

$$Y_{eq}[k] = C[k]\,Y[k] \tag{2.3}$$

where $C[k]$ denotes the complex equalizer for the k^{th} sub-channel. The data symbols are subsequently recovered.

If the ideal low-pass filtered I and Q signal are denoted by $y_i(t)$ and $y_q(t)$, the signals in presence of IQ mismatch are given by

$$\hat{y}_i(t) = y_i(t) \tag{2.4}$$

$$\hat{y}_q(t) = (1+\varepsilon)(y_q(t)\cos(\theta) - y_i(t)\sin(\theta)). \tag{2.5}$$

where ε and θ are gain and phase imbalance parameters as shown in Fig. 1. By taking the discrete Fourier transform of the complex samples, it is readily verified that the sample at the k^{th} sub-carrier in the OFDM receiver will now become:

$$\hat{Y}[k] = \gamma\ Y[k] + \lambda\ Y^*[-k] \tag{2.6}$$

where

$$\gamma = 0.5\{1 + (1+\varepsilon)(\cos(\theta) - j\sin(\theta))\} \tag{2.7}$$

$$\lambda = 0.5\{1 - (1+\varepsilon)(\cos(\theta) + j\sin(\theta))\} \tag{2.8}$$

Equation (2.5) shows that IQ imbalance causes an interference term in the Q channel which is proportional to the in-phase signal. In addition, the quadrature signal amplitude is scaled. In the frequency domain, it is observed in Equation (2.7) that the sample at sub-carrier k is multiplied by a complex factor γ. In addition, a spurious component will be present which is equal to the conjugate of the symbol at $-k$ sub-carrier multiplied by another complex term, λ. The symbol at the k^{th} sub-carrier, therefore, will include an interference related to the symbol at the $-k^{\text{th}}$ sub-carrier, and vice versa.

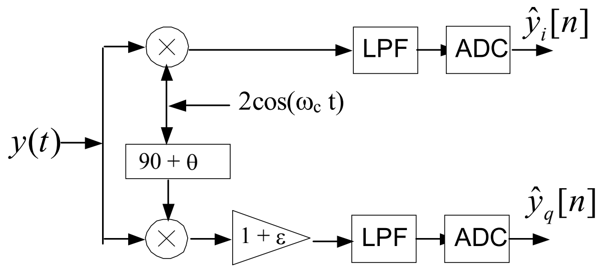

Fig. 1: IQ imbalance in the receiver

III. Time Domain Adaptation

The adaptation algorithm proposed in [3] is based on the effect of IQ mismatch on the baseband time-domain samples. The intent is to predict the interference signal in Q path and subtract it from the incoming signal. This is possible if the power of the interfering signal is smaller than the power of the signal of interest, which is indeed expected to be the case.

A block diagram of the structure of the compensation block is shown in Fig. 3. The mismatch cancellation is done in two stages: first an adaptive filter predicts the interference from the I path and subtracts it from the quadrature signal. Then, the gain in the quadrature path is adjusted adaptively.

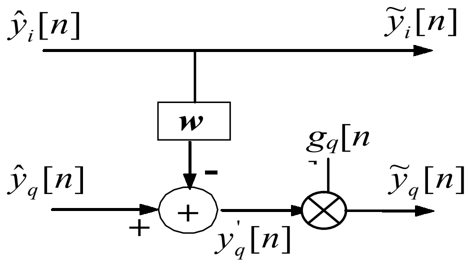

Fig. 2. Time domain compensation block

Let us define $y'_q[n]$ as the sample in the quadrature path after the interference from the in-phase signal is subtracted, i.e.,

$$y'_q[n] = \hat{y}_q[n] - \mathbf{w}^T \hat{\mathbf{y}}_i[n] \tag{3.1}$$

where $\mathbf{w}$ is the filter coefficient vector and $\hat{\mathbf{y}}_i[n]$ is the vector of input samples, i.e.,

$\mathbf{w}^T = (w_0, w_1, ..., w_{m-1})$,

$\hat{\mathbf{y}}_i^T[n] = (\hat{y}_i[n], \hat{y}_i[n-1], ..., \hat{y}_i[n-m+1])$.

From Equations (2.5), it is evident that the phase imbalance causes each sample of the quadrature signal to be affected by the sample in the in-phase path at the same time interval. With the interference thus being memoryless, it is sufficient to employ a prediction filter with only one tap. If the expected prediction error power is defined as

$$J_q = E\{|y'_q[n]|^2\} \tag{3.2}$$

then the update equation for the filter coefficient using the least mean square (LMS) algorithm is given by

$$w(n+1) = w(n) + \mu\, \hat{y}_i[n].y'_q[n] \tag{3.4}$$

where μ is the learning constant [6].

Next, the gain parameter is adapted so that the signal power for the Q path becomes a prescribed fixed value. Let us consider that the target power is given by (P_s + P_n), where P_s is the desired signal power and P_n is the noise power. Also, $e_q[n]$ is defined as the difference between the desired power level and instantaneous power at time n, i.e.,

$$e_q[n] = P_s + P_n - |y'_q[n]|^2 \tag{3.5}$$

The update equation for the gain parameter to minimize $e_q[n]$ is readily obtained as given below:

$$g_q[n+1] = g_q[n] + \delta\ e_q[n] \tag{3.6}$$

where δ is the step size.

IV. Frequency Domain Adaptive Compensation

The frequency-domain adaptive algorithm uses two taps for each channel so that the effect of channel and IQ imbalance are compensated simultaneously [5]. Specifically, an augmented tap is utilized to predict the interference signal from the $-k^{\text{th}}$ channel onto channel k. Two-tap equalization for the k^{th} and $-k^{\text{th}}$ sub-channels is depicted in Fig. 3.

The adaptation algorithm works by minimizing the square of an error signal defined as

$$E^l[k] = D^l[k] - \tilde{Y}^l[k] \tag{4.1}$$

where $D^l[k]$ is the desired demodulated symbol of the k^{th} sub-channel for the l^{th} OFDM symbol, and $\tilde{Y}^l[k]$ is the corresponding equalized sample, which is given by

$$\tilde{Y}^l[k] = \mathbf{C}^{l^H}[k]\,\hat{\mathbf{y}}^l[k], \tag{4.2}$$

where $\mathbf{C}^l[k]$ and $\hat{\mathbf{y}}^l[k]$ are the coefficient vector and the input vector, respectively, i.e.,

$\mathbf{C}^l[k] = (C_1^l[k], C_2^l[k])^T$,

$\hat{\mathbf{y}}^l[k] = (\hat{Y}^l[k], \hat{Y}^{l*}[-k])^T$.

The equalizer taps for each sub-channel are adapted to minimize the mean square error defined in Equation (4.1) using the LMS algorithm, i.e.,

$$\mathbf{C}^{l+1}[k] = \mathbf{C}^l[k] + \nu\ E^l[k]\,\hat{\mathbf{y}}^l[k] \tag{4.4}$$

where ν is the adaptation step size [6].

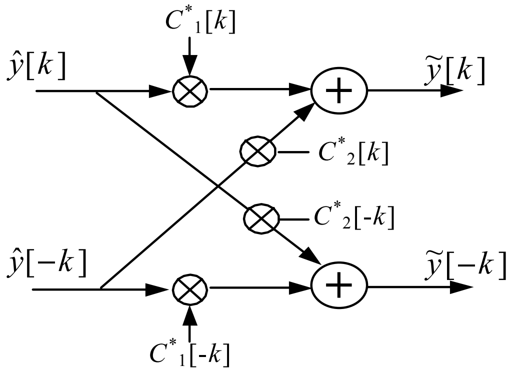

Fig. 3: Two-tap frequency domain equalization (shown for the k^{th} and $-k^{\text{th}}$ sub-channels)

V. Results And Discussion

To evaluate and compare the performance of the two adaptive IQ mismatch cancellation techniques described in previous sections, an OFDM communication system was modeled and simulated. System parameters were set based on the specification of IEEE 802.11a standard for wireless LANs [1]. The standard calls for the use of 48 sub-carriers and 4 pilots. With a null at the DC as well as at 11 other remaining sub-carriers, a 64-point block is formed at the input to the IFFT in the transmitter path. A 16-QAM sub-carrier modulation is chosen for the simulations in this work. An additive white Gaussian noise channel with a flat frequency response is used in the simulations.

To be able to compare the convergence rate of the two techniques, we define the total error power for the l^{th} OFDM symbol as follows:

$$T^l = \sum_{k=-K}^{K} |D^l[k] - \tilde{Y}^l[k]|^2 \tag{5.1}$$

In Fig 4, the learning curves in terms of total error defined above versus the OFDM symbol number for both algorithms are shown when gain and phase mismatches were set at 1.5 dB and 10 degrees, respectively. For each case, the learning constant was set to $1/2\sigma^2$ where σ^2 is the variance of input data to the filter and M is the number of filter taps. Channel signal-to-noise-ratio (defined as the ratio of bit power to noise power) was set at 35 dB. For the frequency domain technique, the true transmitted symbols were used as desired values. Fig. 4 shows that the frequency domain algorithm converges faster than the time domain technique. However, the steady-state total error is much smaller for the latter algorithm. Specifically, it is observed that the steady state error is 3 dB worse for the frequency domain case.

Fig. 5 shows plots of symbol error rate vs. SNR for for the same mismatch settings. The plot for the ideal 16-QAM modulation is also shown for comparison. Notice that when no compensation is applied, IQ mismatch degrades the system performance quite considerably. For example, at SNR=14 dB, the SER is increased by three orders of magnitude. The two adaptive algorithms provide significant improvements. However, while frequency-domain compensation only partially compensates the mismatch effects, the time-domain technique cancels non-ideal effects almost entirely. Specifically, at 14 dB, the SER is still almost two orders of magnitude worse than the ideal case if the frequency-domain approach is employed. The time-domain technique, on the other hand, basically achieves the same error rate as the ideal case.

The disparity between the performance of the two techniques can be partly attributed to the large number of taps added in the frequency domain approach to cancel out interference components due to phase mismatch. With one coefficient added for each sub-channel, in general, N independent *complex* coefficients are utilized. However, based on Equation (2.6), the multiplicative coefficient λ is the same for all sub-channels. The adaptation mis-adjustment for all channels would combine to degrade the SER performance. In contrast, the time-domain technique uses a single multiplier to remove phase effects and an additional tap to cancel our gain mismatch.

A couple of additional points when comparing the two algorithms are in order. First, notice that the frequency domain technique requires a training sequence while the time-domain method works in a blind way. Secondly, for a frequency selective fading channel, the interference term arising from the $-k^{th}$ sub-channel depends on the channel response. Therefore, to apply the two-tap equalization technique described in Section IV, an accurate estimate of the channel response needs to be acquired prior to the operation of the adaptive algorithm. The operation of the time-domain technique, on the other hand, is not affected by the channel fading response.

VI. Conclusions

The performance of OFDM communication systems is severely degraded due to gain and phase mismatches in the mixers. In this paper, two adaptive algorithms for the estimation and compensation such effects in the receiver mixers in an OFDM system were investigated. Simulation results show that while the frequency-domain method converges faster, the time domain approach cancels IQ mismatch almost entirely. In addition, the latter technique does not require training and does not depend on the channel frequency response.

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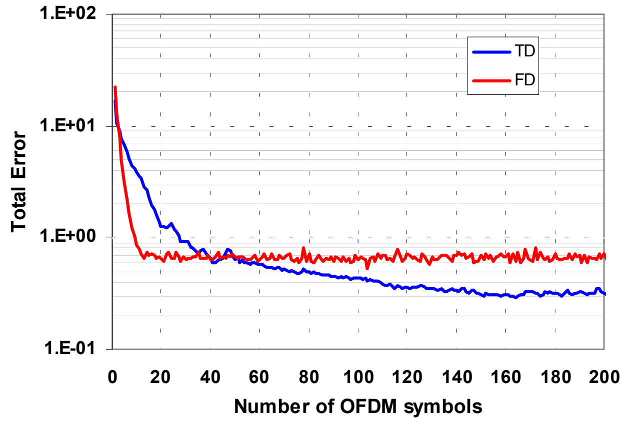

Fig. 4: Learning curves of adaptive compensation algorithms

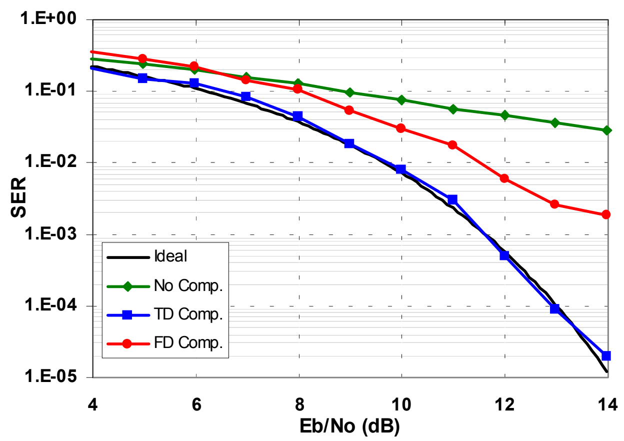

Fig. 5. SER performance of compensation algorithms