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Adaptive I/Q Mismatch Compensation for Wideband Receiver

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Adaptive I/Q Mismatch Compensation for
Wideband Receiver

A thesis submitted in partial fulfillment of the
Requirement for the degree of
Master of Science in Engineering

By

Linda Zhu
B.S, Electrical Engineering, 2004

2014
Wright State University
I HEREBY RECOMMEND THAT THE THESIS PREPARED UNDER MY SUPERVISION
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Abstract

Wide working bandwidth is one of the main concerns in digital wideband receiver. The traditional digital receiver covers only one Nyquist zone, which bandwidth range is from DC to half of the sampling frequency. By utilizing in-phase/quadrature (I/Q) channels, wideband receiver is able to double the working bandwidth, which covers from DC to half of the sampling frequency and also from negative half of the sampling frequency to DC. However, I/Q mismatch in reality introduces unwanted signals, which significantly reduce the system performance and the quality of the received signals.

In this thesis, an adaptive I/Q mismatch compensation technique is presented. A finite impulse response filter is developed, and then the filter coefficients are further optimized to suppress the image signal. The novel and feasible adaptive digital signal processing method has been found to suppress the image signal by 46 – 59.84 dB in comparison with that of the original I/Q channel.
# Table of Contents

Chapter 1 Introduction ............................................................................................................. 1  
  1.1 In-phase (I) and Quadrature (Q) Signals ................................................................. 1  
  1.2 What is I/Q mismatch and compensation?............................................................. 2  
  1.3 Research Motivation ............................................................................................... 4  
  1.4 Thesis Organization ................................................................................................. 4  
Chapter 2 I/Q mismatch Compensation ......................................................................................... 5  
  2.1 I/Q Mismatch Effects in Wideband Receiver ......................................................... 5  
  2.2 Phase Mismatch .................................................................................................... 6  
  2.3 Amplitude Mismatch ............................................................................................. 11  
  2.4 Past work ............................................................................................................. 15  
  2.5 Adaptive I/Q Mismatch Compensation ................................................................. 15  
    2.5.1 Theory ........................................................................................................ 16  
    2.5.2 Architecture and Control Flow ..................................................................... 18  
Chapter 3 Experimental Results ............................................................................................... 22  
  3.1 Matlab Flow ........................................................................................................... 22  
  3.2. Hardware Implementation ..................................................................................... 34  
Chapter 4 Conclusion and Future works ................................................................................. 40  
  4.1 Conclusion ............................................................................................................ 40  
  4.2 Future Works ......................................................................................................... 41  
References ............................................................................................................................... 42
List of Figures

Figure 1.1 Primary and Conjugate Nyquist Zone (same as Second Nyquist Zone) ............ 2
Figure 1.2 Digital Wideband Receiver ........................................................................... 3
Figure 2.1 I/Q Channel FFT based receiver ................................................................ 5
Figure 2.2 Image suppression for Phase Mismatch at 0.1 degree ................................. 8
Figure 2.3 Image suppression for Phase Mismatch at 0.9 degree ................................. 9
Figure 2.4 Image suppression for Phase Mismatch from 0.001 to 0.9 degree ............... 10
Figure 2.5 Image suppression for Amplitude Mismatch at 1.01 .................................. 12
Figure 2.6 Image suppression for Amplitude Mismatch at 1.09 .................................. 13
Figure 2.7 Image suppression for Amplitude Mismatch from 1.001 to 1.09 ............... 14
Figure 2.8 Adaptive Filter Model .............................................................................. 16
Figure 2.9 Digital Input and Output ............................................................................ 17
Figure 2.10 Schematic of System Modeling ................................................................. 18
Figure 3.1 I/Q Compensation Matlab Flow ................................................................. 26
Figure 3.2 Phase and Amplitude Mismatch .................................................................. 27
Figure 3.3 Power Spectrum before and after FIR filter ................................................. 28
Figure 3.4 The Power Spectrum Before and After Adaptive filter. F1 signal has maximum Suppression after Adaptive Filter ................................................................. 29
Figure 3.5 The Power Spectrum Before and After Adaptive filter. F2 signal has Maximum Suppression after Adaptive Filter ................................................................. 30
Figure 3.6 The Power Spectrum Before and After Adaptive Filter for Both Signals ....... 31
Figure 3.7 Dynamic Range for two signals with F1 fixed ............................................ 32
Figure 3.8 Suppression after Original and Adaptive Filter for two signals with F1 fixed... 33
Figure 3.9 I/Q Compensation Hardware Flow ........................................................................36
Figure 3.10 I/Q Compensation with 3 Digital Fractional.........................................................37
Figure 3.11 I/Q Compensation with 6 Digit Fractional ...........................................................38
Figure 3.12 Proposed Receiver Design ..................................................................................39
List of Tables

Table 1: The effect of suppressed image for phase mismatch = 0.1 to 0.9 degree .................. 7
Table 2: The effect of suppressed image for amplitude mismatch = 1.01 to 1.09 ............... 11
Table 3: Image Suppression after adaptive with consideration of both signals .................... 40
Chapter 1 Introduction

Wide bandwidth is one of the most highly desired requirements in the design of the digital wideband receiver. By sampling theorem, the working bandwidth is restricted by the sampling frequency of analogue-to-digital converter (ADC). This bandwidth is considered to be Nyquist zone. In a wideband digital receiver, when using only real-valued input signals, frequencies near Nyquist zone edges are usually not included in the signal detection, for the reason that the working spectrum suffers from aliasing effects. Aliasing causes ambiguity in digital signal processing and makes it impossible to detect the true frequency of the sampled signal data. Consequently, the working bandwidth of digital receiver with real-input signal is smaller than Nyquist zone.

1.1 In-phase (I) and Quadrature (Q) Signals

In-phase (I) and Quadrature (Q) signals were introduced to solve this problem. Traditionally, the bandwidth of the spectrum only covers one Nyquist zone (primary) as mentioned above, with the bandwidth ranges from DC to half of the sampling frequency. By converting input signals of real data into complex data, the bandwidth of the spectrum is able to cover one additional Nyquist zones, the secondary Nyquist zone, with the bandwidth ranges from half of the sampling frequency to the sample frequency. The doubled working bandwidth benefits the receiver in many ways. For example, it also provides better frequency resolution, larger working dynamic range and increases the stability of the system.
Applying fast Fourier transform (FFT) on the complex data, the secondary Nyquist zone is mapped to the negative frequency range, or the “conjugated Nyquist zone, which the bandwidth ranges from negative half of the sampling frequency to DC. Figure 1.1 below displays the name of different Nyquist zones.

![Power Spectrum diagram](image)

**Figure 1.1** Primary and Conjugate Nyquist Zone (same as Second Nyquist Zone)

1.2 What is I/Q mismatch and compensation?

Figure 1.2 shows the general structure of a digital wideband receiver. Basically, when a real-valued incoming signal enters the system, the Preselection Filter selects the signal in the right frequency and remove the unwanted signals. LNA then amplifies the signals and distinguish the signals from the noise. IR filter will again filter out the useful signals and send them to Local Oscillator (LO). The function of LO is to generate the
correct resulting receiver frequency. For instance, if the incoming frequency is 2.56 GHz and we want the output frequency of 0.56 GHz, we can adjust the frequency of LO to be 2 GHz to get the desired receiver frequency. Channel Select Filter, then filter out the useful signals.

As the signals are split and fed into two channels, one channel of signal remains unchanged and the other channel of signal experiences a 90° phase shift. Ideally, I and Q signals have the same amplitude and a phase difference of 90 degrees; however, when this condition is not met, I/Q mismatch arises. The mismatch occurs over every analogue part of the channels, such as local oscillator (LO), mixer, anti-aliasing filter and analogue-to-digital converter (ADC). I/Q mismatch can cause the image signal to interfere with the weak signal, thus reducing the dynamic range. In consequence, I/Q mismatch can cause significant system degradations in a wideband receiver; the effect can be dramatic, therefore, compensation for I/Q mismatch is needed. The compensation of I/Q mismatch is exactly equivalent to cancel the image signal, or minimize the errors due to I/Q mismatch.

Figure 1.2 Digital Wideband Receiver
1.3 Research Motivation

Motivated by the increasing demand of improving I/Q mismatch compensation in wideband receiver and the disadvantage of the existing methods, this research presents a simple adaptive algorithm method to compensate the I/Q mismatch due to the mismatch errors between In-phase and quadrature path. We make efforts to improve the image reduction by 45 to 59 dB.

Both the experiment and simulation results will be demonstrated to confirm the feasibility of the adaptive algorithm. Finally, this research also presents that the developed algorithm can be implemented for real time operation.

1.4 Thesis Organization

This thesis is organized as follows. Chapter 1 gives introduction on I/Q mismatch and the motivation of this research. Chapter 2 explains in detail how I/Q mismatch effects the wideband receiver. In addition, this chapter also introduces the feasible adaptive algorithm to compensate the image errors due to I/Q mismatch. Chapter 3 shows the experimental and simulation results, followed by conclusion of the thesis and discusses the future works in Chapter 4.
Chapter 2 I/Q mismatch Compensation

Figure 2.1 I/Q Channel FFT based receiver

2.1 I/Q Mismatch Effects in Wideband Receiver

In an ideal I/Q channel FFT based receiver as shown in figure 2.1, the incoming signal is multiplied by $\cos(\omega t)$ and $\sin(\omega t)$, it is worth mentioning that the cosine signal refers to as in-phase signal and the phase shift signal refers to as quadrature signal in wideband receiver. The complex exponential expression of the input is

$$e^{j\omega t} = \cos(\omega t) + j\sin(\omega t) \quad (1)$$

where $\omega = 2\pi f$. When there is no I/Q mismatch in the system, the image frequency components cancel each other. Only the signal frequency components are left. However, when I/Q mismatch is considered, the complex number for the output is

$$\cos(\omega t) + j(1+k)\sin(\omega t + \phi) \quad (2)$$
We assume here cosine path is perfect and sine path has all the mismatch. Equation (2) is equivalent to

$$\frac{1}{2}[e^{jwt}(1 + (1 + k)e^{j\varphi})] + \frac{1}{2}[e^{-jwt}(1 - (1 + k)e^{-j\varphi})]$$

(3)

where $k$ is the amplitude mismatch, and $\varphi$ the phase mismatch. The amplitude and phase mismatch creates an ‘image’ signal. This image signal will interfere with the weak signals in wideband receiver. As a result, the receiver’s signal dynamic range will be reduced significantly.

### 2.2 Phase Mismatch

If the phase shift is not exactly 90°, then a phase imbalance is introduced. According to article [2], an equation is given to calculate the image suppression when there is a phase mismatch in I/Q signals.

$$Suppression \ (dB) = 20\log\sqrt{\frac{A^2-2A\cos\varphi+1}{A^2+2A\cos\varphi+1}}$$ \ (4)

where $A$ is ratio of amplitudes of the input signals and $\varphi$ is the phase mismatch.

Suppose $A = 1$ and $\varphi = 0$, the image suppression would likely be -100dB. Now, we set $A=1$ and $\varphi=0.1^\circ$, then the suppression

$$Suppression \ (dB) = 20\log\sqrt{\frac{A^2-2A\cos\varphi+1}{A^2+2A\cos\varphi+1}}$$

$$= -61.18 \ dB$$
The suppression readings have been investigated and calculated, as the phase mismatch increases from 0.1° to 0.9° while the amplitude mismatch remains the same. The values are tabulated in table 1.

Table 1: The effect of suppressed image for phase mismatch = 0.1 to 0.9 degree

<table>
<thead>
<tr>
<th>Phase mismatch</th>
<th>Image suppression</th>
</tr>
</thead>
<tbody>
<tr>
<td>0.1°</td>
<td>-61.18</td>
</tr>
<tr>
<td>0.2°</td>
<td>-55.16</td>
</tr>
<tr>
<td>0.3°</td>
<td>-51.64</td>
</tr>
<tr>
<td>0.4°</td>
<td>-49.14</td>
</tr>
<tr>
<td>0.5°</td>
<td>-47.20</td>
</tr>
<tr>
<td>0.6°</td>
<td>-45.62</td>
</tr>
<tr>
<td>0.7°</td>
<td>-44.28</td>
</tr>
<tr>
<td>0.8°</td>
<td>-43.12</td>
</tr>
<tr>
<td>0.9°</td>
<td>-42.09</td>
</tr>
</tbody>
</table>

Based on the results in table 1, it is obvious to see that as phase mismatch increases, the image reduction decreases. The image suppression equation (4) is verified by the MATLAB program in figure 2.2 and figure 2.3 for phase mismatch equal to 0.1 and 0.9 degree. Figure 2.4 shows the image suppression for phase mismatch for 0.01 to 0.9 degree. The MATLAB results are identical as the calculated results by the equation.
Figure 2.2 Image suppression for Phase Mismatch at 0.1 degree
Figure 2.3 Image suppression for Phase Mismatch at 0.9 degree
Figure 2.4 Image suppression for Phase Mismatch from 0.001 to 0.9 degree
2.3 Amplitude Mismatch

Suppose $A = 1.01$ and $\phi = 0^\circ$, by using the suppression equation (4),

$$Supression\ (dB) = 20\log\sqrt{\frac{A^2 - 2Acos\phi + 1}{A^2 + 2Acos\phi + 1}}$$

$$= -46.06\ dB$$

Table 2 shows all the image suppression for amplitude mismatch from 1.01 to 1.09, when the phase mismatch remains the same. From the table 2, it is obvious to see that as the amplitude mismatch increases, the image reduction decreases.

Table 2: The effect of suppressed image for amplitude mismatch = 1.01 to 1.09

<table>
<thead>
<tr>
<th>Amplitude mismatch</th>
<th>Image suppression</th>
</tr>
</thead>
<tbody>
<tr>
<td>1.01</td>
<td>-46.06</td>
</tr>
<tr>
<td>1.02</td>
<td>-40.09</td>
</tr>
<tr>
<td>1.03</td>
<td>-36.61</td>
</tr>
<tr>
<td>1.04</td>
<td>-34.15</td>
</tr>
<tr>
<td>1.05</td>
<td>-32.26</td>
</tr>
<tr>
<td>1.06</td>
<td>-30.71</td>
</tr>
<tr>
<td>1.07</td>
<td>-29.42</td>
</tr>
<tr>
<td>1.08</td>
<td>-28.30</td>
</tr>
<tr>
<td>1.09</td>
<td>-27.32</td>
</tr>
</tbody>
</table>
Again, this result is verified by the MATLAB program in figure 2.5 for Amplitude equals to 1.01, while phase is 0 degree. Figure 2.6 shows the image suppression for amplitude equals to 1.09 degree. Figure 2.7 displays the image suppression when amplitude mismatch is 1.001, also the mismatch from 1.01 to 1.09.

Figure 2.5 Image suppression for Amplitude Mismatch at 1.01
Figure 2.6 Image suppression for Amplitude Mismatch at 1.09

X: -49
Y: -27.32
Figure 2.7 Image suppression for Amplitude Mismatch from 1.001 to 1.09
2.4 Past work

I/Q mismatch is a big concern in the design of wideband receiver. The topic of I/Q mismatch and the compensation methods have been studied and presented in the past few years [1-19]. The FIR filter derivation in [1] was used as a starting point of this thesis. In the paper, a finite impulse response filter was used to compensate the I/Q mismatch, although the results were not most optimized. I/Q imbalance compensation methods were presented in [1], [18], and [19]. However, the paper only showed narrowband signals rather than the wideband signals. The compensation methods in quadrature receivers with frequency dependent were introduced in paper [5-6], [8], and [17], the problem is that they do not show the input signals coming from both Nyquist zone simultaneously.

2.5 Adaptive I/Q Mismatch Compensation

The method of adaptive I/Q mismatch compensation is presented to compensate the image power due to I/Q mismatch. It is a method for adjusting the adaptive filter coefficients and finding the optimum solution on the performance. We choose this adaptive algorithm, because of its simplicity, stability, and easily implemented.
2.5.1 Theory

A general adaptive filter model is depicted in figure 2.8. The complex output $y[n]$ is used as the reference of the algorithm.

$$y(n) = y_R(n) + jy_I(n)$$  \hspace{1cm} (5)

The filter output can be written as

$$y(n) = \omega^T(n) \ast x(n)$$  \hspace{1cm} (6)

The final error signal output is

$$e(n) = d(n) - x^T(n) \ast w(n)$$  \hspace{1cm} (7)

where $d(n)$ is the desired output. The coefficient adaptation is

$$w(k+1) = w(k) + \mu e(n)x(n)$$  \hspace{1cm} (8)
where $\mu$ is the stepsize which controls how far we move along the error function at each update step. The coefficients of the adaptive filter are $w(k)$.

$$w(k) = w_R(k) + jw_I(k) \quad (9)$$

where

$$w_R(k) = [w_R^{(k)}(0)w_R^{(k)}(1) \ldots w_R^{(k)}(L - 1)] \quad (10)$$

$$w_I(k) = [w_I^{(k)}(0)w_I^{(k)}(1) \ldots w_I^{(k)}(L - 1)] \quad (11)$$

$L$ is the order of the adaptive filter. Applying this concept to our original FIR filter coefficients depicts in the following figure 2.9. A feedback loop is created and it would adjust the filter coefficients to search for the most optimal image reduction.

![Figure 2.9 Digital Input and Output](image-url)
2.5.2 Architecture and Control Flow

By using [1] as a starting point, the authors proposed FIR filter techniques to compensate I/Q mismatch. Figure 2.10 shows the schematic of system modeling.

![Schematic of System Modeling](image)

This figure is the system model that illustrates the imbalance mitigation scheme. The input signal, $X$ with an angular frequency of $\omega=2\pi f$, where $f$ is the input frequency, and it is modeled as a vector with two elements (in-phase and quadrature). After passing through I/Q module, $A$, the output is $Y$ with embedded imbalance signals and $Y$ can be represented as

$$Y = AX$$  \hspace{1cm} (12)

Where

$$X = [\cos(2\pi ft) \hspace{0.5cm} \sin(2\pi ft)]^T$$  \hspace{1cm} (13)

$$Y = [\cos(2\pi ft) \hspace{0.5cm} \gamma \sin(2\pi ft + \phi)]^T$$

From here, the matrix $A$ can be easily derived as

$$A = \begin{bmatrix} A_{11} & A_{12} \\ A_{13} & A_{22} \end{bmatrix},$$  \hspace{1cm} (14)
\[ A_{11} = 1; A_{12} = 0; A_{21} = \gamma \sin(\varphi); A_{22} = \gamma \cos(\varphi) \]

where \( \gamma \) is amplitude mismatch and \( \varphi \) is phase mismatch at frequency, \( f \).

In order to compensate the amplitude and phase mismatch from \( Y \), an imbalance mitigation matrix \( B \), which is the inverse of matrix \( A \), is presented in the model. After passing through \( B \), the balanced signal is restored at output \( X \) and \( X \) can be represented as

\[ X = BY \quad (15) \]

\( B \) can be derived as

\[ B = \begin{bmatrix} B_{11} & B_{12} \\ B_{21} & B_{22} \end{bmatrix}, \quad (16) \]

where \( B_{11} = 1; B_{12} = 0; B_{21} = -\tan(\varphi); B_{22} = 1/\gamma \cos(\varphi) \).

It is important to define the quasi-transfer function of the imbalance mitigation system based on complex-valued representation. Since the input signal of the imbalance mitigation system, \( Y \) is a vector that contains two elements, \( I_{in} \) and \( Q_{in} \). Similarly, the output signal of the imbalance mitigation system, \( X \) is a vector that contains two elements, \( I_{out} \) and \( Q_{out} \). The input and the output can be rewrite as follows,

\[ Y = [I_{in} \quad Q_{in}]^T \text{ and } X = [I_{out} \quad Q_{out}]^T \quad (17) \]

where the I and Q with subscripts ‘in’ and ‘out’ are the in-phase and quadrature components, respectively. If \( Z_{in} \) and \( Z_{out} \) is the input and output signals of the system \( B \), the complex-valued representation would be \( Z_{in} = I_{in} + jQ_{in} \) and \( Z_{out} = I_{out} + jQ_{out} \). The \( Y \) and \( X \) in (17) can then be re-written as

\[ Y = \begin{bmatrix} \frac{1}{2} (Z_{in} + Z_{in}^*) \\ \frac{1}{2} (Z_{in} - Z_{in}^*) \end{bmatrix}^T \quad (18) \]
where $Z_{\text{in}}^\ast$ is the conjugate of $Z_{\text{in}}$, and C and D are given by

$$C = \frac{(B_{11} - jB_{12} + jB_{21} + B_{22})}{2}$$  \hspace{1cm} (21)$$

$$D = \frac{(B_{11} + jB_{12} + jB_{21} - B_{22})}{2}$$  \hspace{1cm} (22)$$

Where $B_{11}$, $B_{12}$, $B_{21}$, and $B_{22}$ are given in (16). The number C and D depend on the phase and amplitude mismatch at the frequency, $f$.

Equation (20) is the ‘quasi-transfer’ function between the imbalanced input signal and the balanced out complex signal in frequency domain. It is called ‘quasi-transfer’ because the input is a combination of the original signals and its image signals. The image signals are practically small compared to the input signals. This fact is helpful as the goal is to design FIR filter for image signal suppression. The output of the imbalanced mitigation system is written as two input FIR series:

$$Z_{\text{out},k} = \sum_{m=-M}^{M} c_m z_{\text{in},(k-m)} + \sum_{m=-M}^{M} d_m z_{\text{in},(k+m)}$$  \hspace{1cm} (23)$$

The first term of the right hand side of equation (23) is from the un-conjugated input signal in (20), and the second term is from the conjugated input signal in (20). Due to the conjugated input, the second term has positive time index in the summation instead of negative seen in the first term. $(2M+1)$ is the tap number of the FIR filters, and the series of $c$ and $d$ are Fourier pairs of $C$ (21) and $D$ (22), respectively.
A FIR filter was developed to suppress the image signals to about 43 dB. We basically extend the work by modifying the existing FIR filter from +2% to -2%. After creating new matrix with every possible combination of the percentage change for the filter coefficients, we then convolve the modified adaptive filter coefficients with the input signals. The desired set of filter coefficients has been found to improve the image reduction by 8.36 dB more for the first signal and 8.44 dB more for second signal. However, this is inconclusive. We need to choose the right set of adaptive filter coefficients for different frequency every time to achieve the most optimum image reduction.
Chapter 3 Experimental Results

3.1 Matlab Flow

Figure 3.1 shows the MATLAB flow for this research. A signal with sampling frequency of 1 GHz is going into the system. In the real-time process, the Hilbert Transform generates the imbalance in-phase and quadrature signals, the imbalance I/Q signals are represented in the real part and imaginary part. The complex form of signals then send to the next block of Cosine and Sine Representation to create the cosine and sine representation of I/Q signals. The in-phase signals are converted to a cosine representation and the quadrature signals are converted to a sine representation. The final step of the operation is to combine both imbalanced cosine and sine signals of I/Q and convolve with the filter coefficients, which produced from the off line process to restore the balance signals.

The first block in the off line process is FFT. It takes the imbalanced I/Q signals from the real-time and calculate the imbalanced phase ($\psi$) and amplitude ($\gamma$). The main role of B matrix system is to filter out the imbalanced I/Q signals. After passing FIR Tap Coefficients Generation, filter coefficients would be generated. As explained in the last paragraph, the filter coefficients convolve with the imbalanced signals from Real-Time process to restore the balance signals.

In MATLAB program, the original FIR filter coefficients were verified as a first step by using the suppression equations in paper [2]. As proved previously, the image suppressions from the MATLAB results in figure 2.4 were exactly same to table 1 for
phase 0.1 degree to 0.9 degree. As for amplitude 1.01 to 1.09, the image suppressions from the MATLAB results in figure 2.7 were identical too to table 2 based on the supp. It is confirmed that FIR filter coefficients were efficient to compensate I/Q mismatch error.

In MATLAB program, 1 GHz is used as sampling frequency. Since FFT has a length of 256 data point, so the bin is 3.906 MHz each. Two simultaneous signals are chosen based on the 256 frequency intervals, one from positive Nyquist zone range from DC to 500MHz and the other from negative Nyquist zone range from -500 MHz to DC, are fed to the system. For example, we chose the first signal to be 51st bin number which has frequency of 195.312500 MHz from the positive Nyquist zone, and the second signal at bin number of 191 with frequency of 742.187500 MHz from the negative Nyquist zone.

The amplitude and phase mismatch spectra are shown in figure 3.2. The phase mismatch is from 2 to 10° in positive zone and from 10 to 2° in conjugate zone. The amplitude mismatch is from 0.9 to 1.1 in positive zone and 1.1 to 0.9 in conjugate zone. The two signals have same signal to noise of 100 dB. The tap number is 11 for the FIR filter.

In figure 3.3, by using the original FIR filter for I/Q mismatch match compensation, the image power of the first signal before mitigation is -26.8 dB. After the mitigation process, the image signal was suppressed down to -64.44 dB, which gives 37.64 dB reductions. The image power of second signal before and after mitigation is -25.52 dB and -76.7 dB, respectively; the reduction is about -51.18dB.

In figure 3.4, we focus to give the first signal from the positive Nyquist Zone the maximum image suppression with adaptive algorithm. The new image power for the first signal after adaptive filter is -121 dB, which gives 56.56 dB more in reduction.
However, the new image power of the second signal from the negative Nyquist zone is -53.94 dB, which was worse than what we had before.

In figure 3.5, we focus on the second signal from the negative Nyquist zone for the maximum image suppression after applying adaptive algorithm. The new image power for the second image signal was -104.7 dB, compared to -76.93 before. The new reduction was 28 dB more. The new image power for the first signal is -69.67, which gives 5 dB more in reduction.

In figure 3.6, we take both signals into considerations for the best image reductions. After adaptive algorithm, the first image signal has reduction of 8.36 dB more with new image suppression of -72.8 dB. The second image signal has new reduction of 8.44 dB more with new image suppression of -85.36 dB, compared to the original FIR image suppression reading of -64.44 dB and -76.7 dB before, respectively.

In figure 3.7, the working dynamic range for both signals is presented. We fixed the first signal and sweep the second signal through the entire frequency from 1 to 1 GHz. The fixed first signal (represented in red) showed a “straight line” in the graph as expected. The suppression was about 37.64 dB for this particular signal. The suppression for the second signal ranges between 22 to 65 dB for different frequencies. However, there are two spots which appear to be out of range. First spot at 0 dB is due to the same frequency to the fixed first signal. The second spot is because the frequency is at the edge of the bandwidth.

Figure 3.8 shows the image suppression after mitigation between original FIR and the adaptive filter coefficients for both signals. The first signal was again fixed and the second signal from 1 to 1 GHz. It was shown that this particular adaptive filter coefficient gives the most optimal suppression results as shown figure 3.5 for the two
signals (195.312500 MHz and 742.187500 MHz) we had chosen, did not work for all frequencies. The fixed first signal showed a new image reduction of 8.36 dB as expected. Nevertheless, the new adaptive filter coefficient did not give optimal results for the frequency range of the second signal. Therefore, we can conclude that the particular adaptive filter coefficient does not give the most optimal image suppression for all frequencies. When dealing with new frequencies, new adaptive filter coefficient needs to be adjusted in order to get the best image reduction.
Figure 3.1 I/Q Compensation Matlab Flow
Figure 3.2 Phase and Amplitude Mismatch

Mismatch in phase

Mismatch in amplitude

Figure 3.2 Phase and Amplitude Mismatch
Figure 3.3 Power Spectrum before and after FIR filter
Figure 3.4 The Power Spectrum Before and After Adaptive filter. F1 signal has maximum Suppression after Adaptive Filter.
Figure 3.5 The Power Spectrum Before and After I/Q Balance Compensation

F2 signal has Maximum Suppression after Adaptive Filter.
Figure 3.6 The Power Spectrum Before and After Adaptive Filter for Both Signals
Figure 3.7 Dynamic Range for two signals with F1 fixed
Figure 3.8 Suppression after Original and Adaptive Filter for two signals with F1 fixed
3.2. Hardware Implementation

Figure 3.9 depicts the proposed I/Q imbalance compensation hardware flow. Each hardware block represents a functional block in I/Q imbalance compensation MATLAB flow in figure 3.1. The one time process is to implement an I/Q imbalance compensation FIR filter with pre-calculated coefficients into a FPGA receiver system. The design is mainly refer to the Real Time Processing Block, which includes Cosine and Sine Representation Generation, Signal Combination, and I/Q Imbalance Compensation Filter. The test signals to the FPGA receiver system are the same ones used to derive the FIR coefficients. The long term goal is to demonstrate the realistic scenario, in which a Hilbert transform is implemented to convert the digitized signal to real and complex domain where the I/Q signals are imbalanced. These imbalanced signals are fed to the Off-line Processing Block to calculate the coefficients of the I/Q imbalance compensation filter.

One difference between the MATLAB algorithm and the hardware algorithm implementation is that the implementation must specify the precision of arithmetic operation. In this research, we are to create an optimized implementation of the algorithm by: 1) conversion to fixed-point arithmetic, 2) round-off error minimization, and 3) performance verification and satisfaction to support this conversion.

One important point to improve hardware implementation is that FIR filter coefficients in the MATLAB simulation have 15-digit decimal precision to minimize round off error. However, this 15-digit decimals will require more memory space than necessary in the digital receiver design. As a result, we can reduce the number of digit to achieve the same results. We tried FIR coefficients using 3-digit fractional number as shown in figure 3.10, image signals from Nyquist Zone 1 are not suppressed enough
compared to the ones from MATLAB simulation. Using 4 and 5 fractional number still cannot give the results close to the MATLAB simulation. Finally, figure 3.11 demonstrates that 6 digit fractional number the FIR filter begins to perform as well as the MATLAB simulation. By using only 6 digits instead of 15, we would be able to save tremendous time in hardware implantation; therefore, 6 digit fraction number is chosen to minimize hardware implementation and optimize its operation.

Figure 3.12 demonstrates a proposed receiver design used in this research to calculate the new adaptive filter coefficients, as the incoming signals are split and fed into the two channels. When the channel switch (Sch) closes, I/Q switch at the adaptive I/Q mismatch (Siq) opens up and make the incoming signals go forward through the rest of the digital signal processing. When the Sch opens up, Siq closes and the incoming signals will be tunable through adaptive I/Q mismatch filter. The new adaptive filter coefficients will be obtained to compensate the error due to the I/Q mismatch.
Figure 3.9 I/Q Compensation Hardware Flow
Figure 3.10 I/Q Compensation with 3 Digital Fractional
Figure 3.11 I/Q Compensation with 6 Digit Fractional
Figure 3.12 Proposed Receiver Design
Chapter 4 Conclusion and Future works

4.1 Conclusion

I/Q mismatch creates an image signal and reduces the working dynamic range. It can also cause large degradation in a wideband receiver. This thesis presented a feasible DSP solution applying adaptive algorithm to compensate I/Q mismatch problems. The simulation results were conducted to confirm the feasibility of the proposed compensation method. With this method, as shown in table 3, the first image signal is able to suppress 8.36 dB in addition to 37.66 dB from the original FIR, a total of 46 dB in reduction. The second signal is able to suppress 8.44 dB in addition to 51.10 dB, a total of -59.84 dB in reduction.

<table>
<thead>
<tr>
<th></th>
<th>Image suppression for F1 (dB)</th>
<th>Image Suppression for F2 (dB)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Before</td>
<td>-26.8</td>
<td>-25.52</td>
</tr>
<tr>
<td>After</td>
<td>-64.44</td>
<td>-76.92</td>
</tr>
<tr>
<td>Adaptive</td>
<td>-72.8</td>
<td>-85.36</td>
</tr>
<tr>
<td>Improved Reduction</td>
<td>-8.36</td>
<td>-8.44</td>
</tr>
<tr>
<td>Total Reduction</td>
<td>-46</td>
<td>-59.84</td>
</tr>
</tbody>
</table>
4.2 Future Works

There are more works we can do to improve the system operation of digital wideband receiver. We need to further implement hardware in FPGA design; in addition, we can consider clock skew and clock jitter in the design. Clock skew is the average delay between the two signals and clock jitter is deviation of the signal edge from the ideal location. Since they are unavoidable in practical digital design, we can make efforts to minimize the errors caused by them in order to improve the performance of the system.
References


