ADAPTIVE IQ MISMATCH CANCELLATION FOR QUADRATURE IF RECEIVERS

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Abstract: *Analog Implementations of the wireless receivers suffers from the Inphase-quadrature phase (IQ) imbalances between the components of the LPF and the LO in the I/Q paths. The gain and phase errors in the local oscillators and the in the filters account for the I/Q errors .Due to these imbalances the performance of the receivers and the quality of the received signal are degraded due to the gain and the phase imbalances in the I/Q phases. Many approaches for the correction of the IQ channel mismatches and the previous methods applied for correction of IQ mismatches using analog and digital domains have been reviewed. The trade-offs and the issues behind the design and implementation of the wireless receivers are described. The different modulation schemes of wireless receivers are also mentioned. Finally, A novel and a feasible adaptive digital signal processing method, using LMS algorithm, which gives a solution for the correcting IQ mismatches in a Direct Conversion Receceiver, has been proposed and implemented. The algorithm that involves the LMS method has been found to improve the SNR by 15-40 dB.*

I INTRODUCTION

Design of the Wireless receivers involving low cost, low power dissipation and smaller power factor have always been an aggressive goal, with the condition that they must supply the usual bandwidth and sensitivity limitations. There exists a trade off between them to meet a certain specification. A quadrature receiver uses two distinct channels to form the in-phase (I) and the quadrature-phase (Q) components of the received signal. Each channel consists of a mixer, low pass filter and A/D converter. The mismatch between the LP filters and the mismatch between the local oscillators in the I and Q paths can severely limit the performance of the adaptive cancellers and the matched filters and hence reducing the quality of the signal. These errors are caused by the amplitude and the phase imbalances in the mixers when there are multiple received channels or in a given mixer overtime when the mixer oscillator is noisy. Noisy transmitter oscillator, unbalanced low pass filters are the main sources of I/Q errors. Mismatches differ with respect to different receivers. Superhetrodyne receivers have a principle issue which is a trade off between the image rejection and adjacent channel suppression. They require image rejection to suppress the image frequency, which is located two intermediate frequencies away from the desired radio frequency. The direct conversion receiver does not suffer from the problem of image rejection and hence it suffers much less from the mismatch-induced effects than do image reject architectures.

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Various approaches for the correction of I/Q channel imbalance have been proposed using the analog and digital methods. Analog solutions are modifying the circuitry [3], careful layout [4] and making the circuit more robust [5], [6]. Digital solutions are using off-line compensation of the channel imbalances [7], [8] using the Hilbert transform to generate I/Q signals in the digital domain [9] and using the delta –sigma modulator instead of the analog to digital converter with an adaptive mismatch cancellation system. In this paper a novel algorithm which compensates the signal-to -noise ratio (SNR) degradation due to mismatch between the I and the Q paths has been proposed. This algorithm uses a robust adaptive filter that uses LMS algorithm.

II. THE I/Q MISMATCH

The In-phase (I) and the Quadrature (Q) channels are necessary for any angle modulated signals because the two sidebands of the RF spectrum contain different information and may result in irreversible corruption if they overlap each other without being separated into two phases. The demodulator at the receiver has to be synchronous in nature. The receiver must possess an oscillator, which is at the exactly same frequency, and phase as the carrier oscillator, at the transmitter end. The low pass filters at the two paths of the receiver must have identical characteristics, as any mismatch in their characteristics would lead to I/Q errors.

In this paper the I/Q mismatch that occurs in the Direct Conversion receivers is mentioned and a feasible DSP solution has been proposed. The direct conversion (Zero IF or Homodyne conversion) receivers converts a signal from RF to Base band. The band of interest is translated to zero and then the signal is low-pass filtered to suppress the interferences. The direct conversion receivers do not suffer from the problem of images as the intermediate frequencies are zero and hence they do not require image reject filters. The main problem faced in the design of the direct conversion receivers is the I/Q mismatch. The errors that occur due to the phase shift between the oscillators and change in the coefficients/phase shift between the low pass filters in the I/Q paths, corrupt the signal to a large extent and severely distort the signal to noise ratio. The gain imbalance appears as a non-unity scale factor in the amplitude while the phase imbalance corrupts one channel with a fraction of data pulses in the other channel. The Fig.1 shows the block diagram of a QPSK (Quadrature phase shift keying) receiver. Assuming no gain or phase imbalance between the I/Q paths then the signal after demodulation passes through the low pass filter and are received as undistorted signal. The amplitude and the phase mismatches are usually random and changes from time to time. A sixth order butter worth low pass filter with cut-off frequency 8.5 Mhz is used for the experimental simulation and analysis.

III. PROPOSED SOLUTION FOR THE I/Q MISMATCH

The aim is to minimize the errors that are caused due to the mismatch that exist between the filters and the oscillators in the two paths. In ideal case the oscillators in the two paths must very stringently oscillate at the same frequency and phase and the low-pass filters in both of the paths must have identical characteristics. Error occurs when there is mismatch (phase or gain imbalance) between the two oscillators and the low-pass filters. Compensation must be done in order to reduce the error. A novel and a feasible adaptive digital signal processing method, which gives a solution for the IQ mismatches, has been proposed. This algorithm involves the LMS method and the SNR have been found to improve by 15-40 dB. Since the two channels contain different signals, the mismatch could never be measured, especially if the mismatch is random. To avoid this difficulty, the two signals in the In-phase and the quadrature phase are made the same and then compared. They are now supposed to be identically equal and a filter using an LMS algorithm is used to compensate for the mismatch.

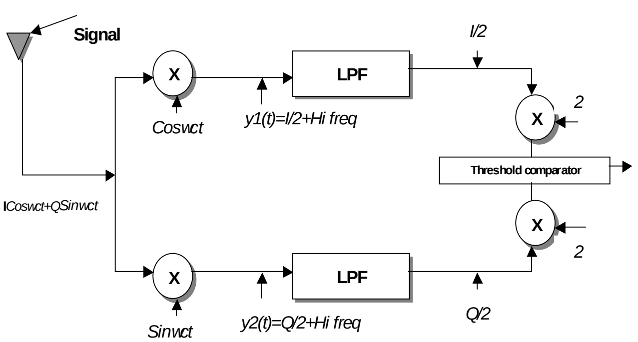


Fig1. QPSK (Quadrature phase shift keying) receiver

IV. COMPENSATION FOR FILTER MISMATCH

The Fig2. shows the circuit for compensating for the filter mismatch. The basic principle is to make the input signal through the filters identical and then compensate for the error between the two outputs. The compensation filter is an adaptive filter using the LMS algorithm to reduce error. The switch S is closed and the switches Sm, S1 and S2 are open. The calibration of the circuit is essentially done with a random input signal. Thus the circuit initially is trained so that the two signals at the filter output would become identical. After training, the weights are calculated and the transfer function of the new filter is obtained and added to the network as shown in Fig 3. The results after mismatch compensation are obtained and tabulated. The SNR, the signal to noise ratio has been found to increase and the errors have been decreased as the LMS filter showed improvement for all kinds of mismatches. The improvement in the SNR due to the both change in the odd coefficients and change in the phase of the filter has been investigated and calculated. The signal to noise ratio decreases (I/Q mismatch increases), as the percentage of change in the odd coefficients of the filter and as the phase change increases (Tables 2&3). The SNR have been found to improve with the increase in the number of filter taps, but the improvement is not significant as the number of taps increase above 30 taps. The improvement in the average values of SNR for % change in the filter coefficients and change in the phase of the LMS filter is tabulated (Table1) for different values of filter taps.

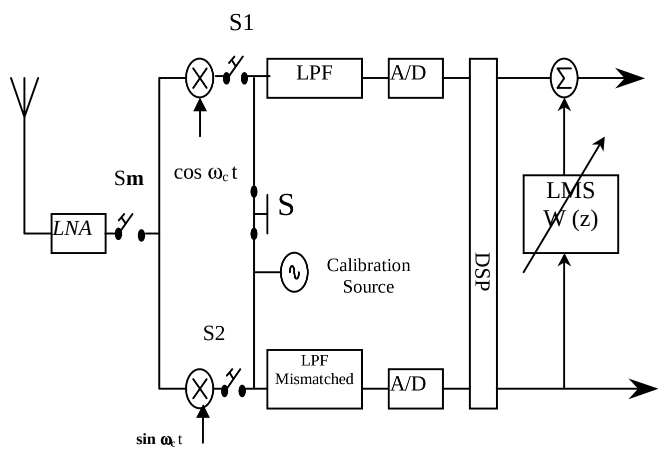


Fig2. Calibration circuits for filter mismatch correction

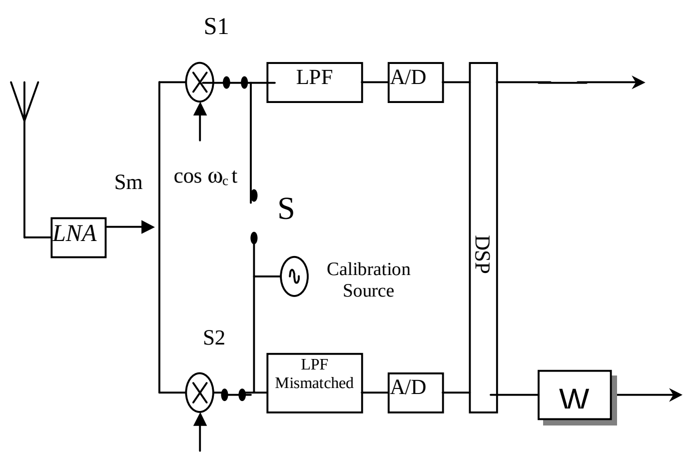


Fig3. Circuits for filter mismatch correction

Average SNR before correction	42.1124 dB

Number Of Filter Taps	Average improvement in SNR (dB)
10	58.4684
20	65.9654
30	68.9870
40	70.4294
50	71.1380

Table(1) Improvement of SNR after correction of Filter Mismatch

% of change in the filter coefficients	SNR before LMS dB	SNR after LMS (10 taps) dB	SNR after LMS (20 Taps) dB	SNR after LMS (30 taps) dB	SNR after LMS (40 taps) dB	SNR after LMS (50 taps) dB
-2.0	34.3058	60.5603	63.7796	64.2808	64.4454	64.4506
1.0	40.4035	56.5062	65.1999	68.4683	69.6306	70.0562
-0.5	46.4647	60.2487	68.6326	72.5734	74.4545	75.3478
0.5	46.5539	59.7720	67.3705	71.4234	73.6099	74.7797
1.0	40.5850	54.3766	61.8397	65.7159	67.7922	68.8966
2.0	34.6983	51.7462	58.2077	61.3299	62.8511	63.6003

Table (2) The effect of the % of change in some of the coefficients of the LPF before and after applying the LMS

% Change In the phase	SNR Before LMS dB	SNR after LMS (10 taps) dB	SNR after LMS (20 Taps) dB	SNR after LMS (30 taps) dB	SNR after LMS (40 taps) dB	SNR after LMS (50 taps) dB
-2.0	46.6191	60.0822	67.9730	71.9802	74.0594	75.1302
-1.0	52.6334	65.9365	73.8203	77.8744	79.9901	81.0851
-0.5	58.6509	71.8810	79.7590	83.8351	85.9680	87.0743
0.5	58.6446	71.7421	79.6052	83.7227	85.8873	87.0146
1.0	52.6208	65.6582	73.5129	77.6496	79.8289	80.9658
2.0	46.5939	59.5226	67.3596	71.5321	73.7382	74.8926

Table (3) The effect of the phase change in the LPF before and after applying the LMS

V.COMPENSATION FOR OSCILLATOR MISMATCH

Another main factor that leads to the I/Q error is the mismatch due to the local oscillator. The circuit for calibration of the local oscillator mismatch is shown in the Fig4. For calibration, the random signal is applied to the local oscillator when the switches S, Sm are open and the switches S1, S2 are closed. Another LMS algorithm similar to the previous case is applied to account for the mismatch. After calibration, the transfer function of the filter is added that compensates for the mismatch caused by the oscillators. The circuit after correction of the local oscillator mismatch is shown in the Fig5. The improvement in the SNR due to change in both the % amplitude and the phase of the local oscillator has been investigated. The signal to noise ratio decreases (I/Q mismatch increases), as the percentage change in the amplitude and the phase of the oscillator increases (Tables 5&6). The SNR has been found to improve with the increase in the number of filter taps, but the improvement is not significant as the number of taps increase above 30 taps.

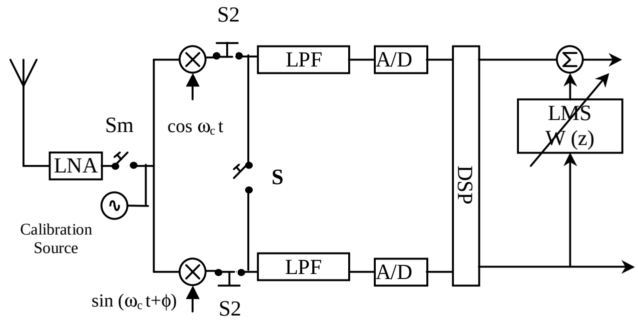


Fig4. Calibration circuits for oscillator mismatch correction

The improvement in the average values of SNR for % change in the amplitude and the phase of the local oscillator is tabulated (Table4) for different values of filter taps.

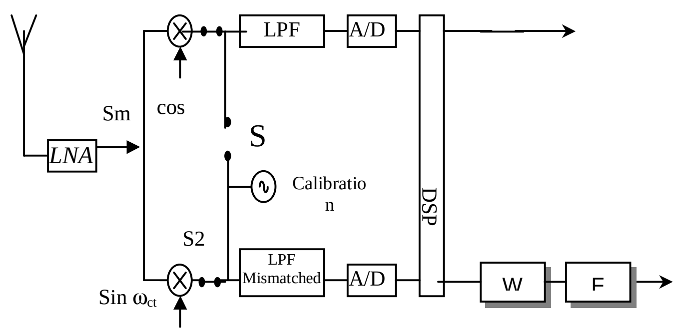


Fig5. Circuits for oscillator and filter mismatch correction

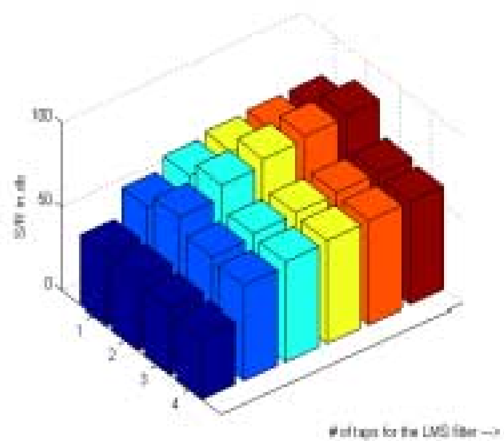
Average SNR before correction 37.7027 dB	
Number Of Filter Taps	AVERAGE SNR dB
10	56.5115
20	59.1093
30	60.3417
40	60.9745
50	61.2873

Table(4) Improvement of SNR after correction of Local Oscillator Mismatch

Table (5) The effect of the phase change in the LO before and after applying the LMS

% of change in the amplitude of the LO	SNR before LMS dB	SNR after LMS (10 taps)	SNR after LMS (20 taps)	SNR after LMS (30 taps)	SNR after LMS (40 taps)	SNR after LMS (50 taps)
-3	36.3469	55.1557	57.7535	58.9859	59.6187	59.9180
-2	39.9127	58.7216	61.3193	62.5517	63.1845	63.4838
-1	45.9771	64.7859	67.3836	68.6161	69.2488	69.5482
1	46.0639	64.8728	67.4705	68.7029	69.3357	69.6351
2	40.0864	58.8953	61.4930	62.7254	63.3582	63.6576
3	36.6075	55.4163	58.0141	59.2465	59.8793	60.1786

Table (6) The effect of the % of change in the amplitude of the LO before and after applying the LMS



1 Effect of change in the filter coeff
2 Effect of change in the filter phase
3 Effect of change in the LO phase
4 Effect of change in the LO amplitude

SNR before using

SNR after applying

SNR after applying

SNR after applying

SNR after applying

Fig.6

Phase change in the LO phase	SNR before LMS dB	SNR after LMS (10 taps)dB	SNR after LMS (20 taps)dB	SNR after LMS (30 taps)dB	SNR after LMS (40 taps)dB	SNR after LMS (50 taps)dB
-3.0	31.7360	49.8774	52.1237	53.1362	53.6485	53.9063
-2.0	35.2262	53.6115	55.9712	57.0485	57.5931	57.8595
-1.0	41.2145	59.8241	62.3019	63.4526	64.0368	64.3166
1.0	41.2145	60.1924	62.9076	64.2290	64.9201	65.2471
2.0	35.2262	54.3380	57.1630	58.5781	59.3376	59.7022
3.0	31.7360	50.9434	53.8645	55.3736	56.2106	56.6243

Fig.6 shows the improvement of the SNR using the entire algorithm developed using the LMS filter.

CONCLUSION:

The paper addresses the I/Q mismatch problems in the direct conversion receivers. The mismatch was initially calibrated and an adaptive approach to match the I and the Q components of the complex valued inputs were presented and the I/Q mismatch were removed by using a compensation filter using an LMS algorithm. This method of I/Q mismatch compensation has been found to remove any deleterious effects that would be caused if there were mismatches between the I/Q phases. Thus the algorithm developed has been found to improve the SNR by 15-40 dB.

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