Frequency Selective IQ Phase and IQ Amplitude Imbalance Adjustments for OFDM Direct Conversion Transmitters

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Abstract – Low power and low cost analog front-end architectures are required to provide competitive applications based on e.g. the IEEE802.11a Wireless LAN standard. Direct conversion analog front-end architectures can fulfill these demands but may introduce unwanted imperfections like IQ amplitude and IQ phase errors. This paper presents a fully digital solution to eliminate frequency selective IQ phase and IQ amplitude imbalance errors caused by the analog modulator in conjunction with low cost analog base-band filters. This paper covers additionally the relevant hardware and software partitioning of the related error detection and error correction blocks.

Index Terms – OFDM, direct conversion, frequency selective IQ phase imbalance, frequency selective IQ amplitude imbalance

I. Introduction

Fully digital compensation techniques for IQ phase and IQ amplitude imbalance errors provide advantages, because they offer a cheap solution with regards to the overall radio architecture. The IQ imbalance errors can appear in the case of a direct conversion analog front-end architecture. These architectures are recommendable if the radio application requires a low cost solution. This is very often true for portable or wireless devices such as IEEE802.11a Wireless LAN applications. To provide anyhow the required high signal accuracy one has to guarantee that the analog direct conversion front-end IQ imperfections will be eliminated satisfactory.

The use of cheap analog front-end components might introduce analog filter amplitude ripple and group delay, which make the IQ imbalance errors frequency selective. The frequency selectivity can be removed successfully by employing frequency selective IQ phase and frequency selective IQ amplitude adjustment algorithms. The here presented time domain adjustment techniques employ a decision directed IQ amplitude and filter pre-equalizer. The non-decision aided IQ phase adjustment provides a pre-equalizer-like architecture and processes a certain amount of time domain coefficients, too.

Both imperfections, the frequency selective IQ amplitude and IQ phase imbalance errors, can be assumed to vary very slowly over a certain period of time. Hence the IQ error detections are required during the tracking phase relatively seldom. In contrast to that the IQ error corrections are needed all the time to correct all outgoing samples continuously. Hence it is possible to implement the IQ error detection algorithms via software on a Digital-Signal-Processor (DSP) and the IQ error corrections via hardware in an ASIC or FPGA.

Based on an example for a direct conversion OFDM transmitter this paper will present the mathematical equations for an IQ phase and IQ amplitude imbalance error generation and the corresponding frequency selective IQ phase and IQ amplitude error adjustments. For both algorithms there will be given an implementation overview and simulation results provide visual comparability.

II. Direct Conversion Front-End

A possible direct conversion analog front end for an IEEE802.11a OFDM transmitter is given by figure 1). Starting from the left upper corner in figure 1) the digital IQ symbols are generated and the IFFT block converts them from the frequency domain to the time domain. The next block takes care about correcting the IQ amplitude and analog filter imperfections via a pre-equalizer. The IQ phase error is pre-compensated by the IQ phase pre-equalizer. Both blocks operate on the ideal incoming time domain symbols and pre-modify these with the corresponding correction values.

After the IQ pre-processing has been finished the digital signal is handed over to the analog domain, where the analog low-pass filters might insert filter amplitude ripple and group delay. The analog base-band I-branch and the analog Q-branch differ physically and hence the analog filters most probably do not equal exactly. Hence the frequency selective I- and Q-imperfections in both branches differ and the correcting pre-equalizers operate as real-number devices in both branches independently. Complex-number pre-equalizers would not be able to provide the required corrections.
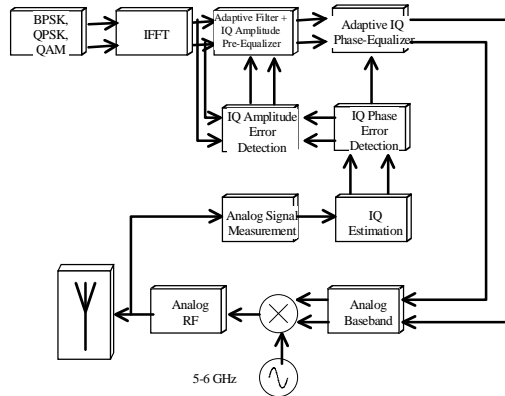

Fig. 1. Example of an IEEE802.11a wireless LAN direct conversion analog front-end.

The analog IQ modulator will add the IQ phase imbalance imperfections. Finally the outgoing signal will be measured at the antenna input port. An analog envelope measurement is fed back to the digital base-band transmitter domain. The envelope can act as the signal source to find out the required information about the IQ imperfections. A digital base-band block provides estimates of the virtual analog I- and Q-values at the antenna input port without a down-modulation process. Because the RF signal is real and not complex, one can only estimate base-band equivalent I- and Q-branch samples. In case of possible estimation errors it can be assumed that the wrong values are negligible during the loops' tracking phase. During the acquisition mode the IQ estimation in conjunction with the IQ feedback loops provide reliable estimates, too. During that time wrong estimates will be low-pass filtered through the feedback loop architecture.

After the analog I- and Q-symbol estimation has been calculated successfully, the frequency selective IQ phase and IQ amplitude error detection take place. The IQ phase error detection is a blind algorithm and requires only the IQ symbol input from the estimation block. The IQ amplitude and filter error detection algorithm needs both, the analog IQ estimates and the corresponding ideal IQ symbols. All information is necessary, because the IQ amplitude and filter pre-equalizer is decision aided. After all errors have been calculated and the correction coefficients for both pre-equalizers have been provided, the feedback loops close with the IQ correction blocks.

The next sections will provide mathematical descriptions, block and implementation diagrams for the IQ phase and IQ amplitude error generation and the error adjustments.

III. IQ Phase Imbalance Error

For the analog mixing process two signals, a sine- and a cosine-signal, have to be provided. Because of technical reasons precise orthogonal functions cannot be guaranteed and hence an additional offset angle $\varphi \neq 0°$ will be measurable between the sine- and cosine-functions. This phenomenon will be called non-frequency selective IQ phase imbalance error.

Additionally the analog base-band components like low-pass filters might add frequency selective imperfections like amplitude ripple or group delay. Hence the non-frequency selective IQ phase imbalance imperfections will result in frequency selective IQ phase imbalance inaccuracies. The non-frequency IQ phase imbalance error is formulated in equation (1).

$$s'(t) = I'(t) + j \cdot Q(t) \qquad\qquad (1)$$
$$= I(t) + Q(t) \cdot \sin(\varphi) + j \cdot Q(t)$$

One can see, that a certain part of the Q-branch, which is proportional to $\sin(\varphi)$, has been added to the I-branch. Hence the ideal, un-correlated I- and Q-branches will be correlated after an IQ phase error has been inserted. Before the IQ phase error insertion in equation (1) takes place imperfect analog filters might introduce frequency selectivity. Hence at the antenna input port it is not possible anymore to divide between the IQ phase imbalance and the analog filter effects. Hence digital pre-compensation techniques need to employ a pre-correcting signal that provides frequency selectivity. This can be achieved by a pre-equalizer-like compensation architecture, which will be presented in this paper.

The IQ phase error insertion normally introduces at the same time a small IQ amplitude error. This additional IQ amplitude error has been neglected in equation (1).

IV. IQ Amplitude Imbalance Error

The IQ amplitude imbalance error is formulated in equation (2).

$$s'(t)=I'(t)+j \cdot Q'(t) = a \cdot I(t)+j \cdot b \cdot Q(t) \qquad (2)$$

Each of the two branches can have an own amplification factor, because analog I-branch and Q-branch components might differ in terms of their amplification. Because the analog filters will introduce amplitude ripple in the pass-band the IQ imbalance errors might become frequency dependent.
A time domain pre-equalizer handles the frequency selective imperfections [4].

The following two sections introduce the frequency selective IQ phase and the frequency selective IQ amplitude adjustment loops.

V. Frequency Selective IQ Phase Imbalance Adjustment

In this section it is assumed that one or both analog base-band filters provide imperfections. These imperfections could be one or more items like amplitude ripple or non-linear filter phase behavior. Because of these additional imperfections a non-frequency selective IQ phase imbalance adjustment loop locks to a wrong error value.

Hence it is necessary to implement an IQ phase imbalance error detector, which is frequency selective and able to cover analog filter imperfections. The $\tilde{I}$ and $\tilde{Q}$ syntax defines I and Q samples which are afflicted by the frequency selectivity of the analog filters.

Equation (3) describes the mathematical operations for the error detection $e_i[n]$. N different IQ phase errors are calculated.

$$e_i[n]= \tilde{I}[n-(N-1)/2] \cdot \tilde{Q}[n-(i-1)] \qquad (3)$$
$$i=1,2,\cdots,N$$
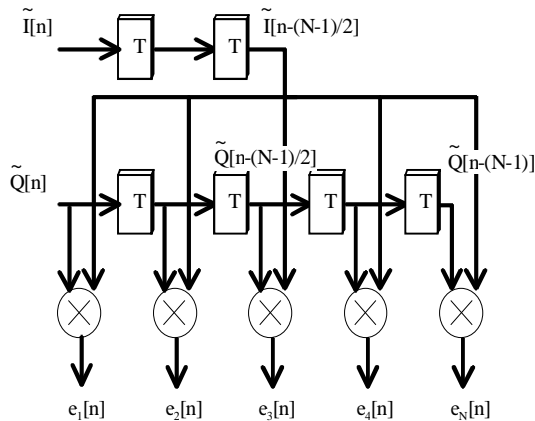
In this paper it is assumed that N is an odd number.

Fig. 2) Frequency selective IQ phase imbalance error detector. N = 5.

Figure 2) presents a possible implementation setup. The center-tap (N-1)/2 of the I-branch will be multiplied with N different values from the Q-branch. Equation (4) shows that each error value $e_i[n]$ will be low-pass filtered by its own integrator.

$$c_i[n]= \mu \cdot \int_{k=0}^{n} e_i[k] \ , \ i=1,\cdots,N \qquad (4)$$

The constant μ describes the step width of the adaptation loop. The final frequency selective IQ phase imbalance error correction will be done by equation (5) and figure 3).

$$I'[n-(N-1)/2]= \begin{bmatrix} I[n-(N-1)/2]- \\ \sum_{i=1}^{N} c_i[n-m] \cdot Q[n-(i-1)] \end{bmatrix} \qquad (5)$$
$$i=1,2,\cdots,N \ and \ m>0$$

The variable m describes the implemented loop latency. Similar to a normal pre-equalizer the Q-branch values from the tap-delay line are multiplied with the corresponding correction coefficients c_i and are summed up. This result is subtracted from the perfect I-branch center tap. Hence the digital I-branch values start to provide a frequency selective IQ phase imbalance error, which will be compensated after the analog IQ phase imbalance error has been inserted. Frequency selectivity is needed to pass the analog base-band filters correctly.
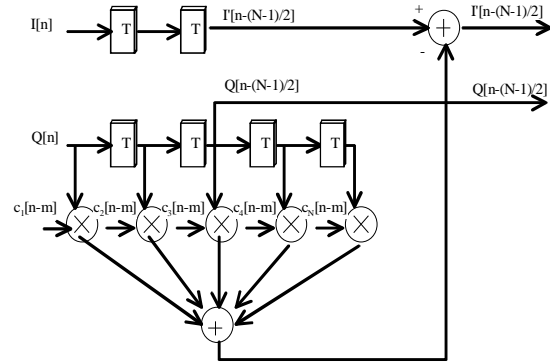

Fig. 3) Frequency selective IQ phase correction. N=5.

VI. Frequency Selective IQ Amplitude Imbalance Adjustment

In this section there will be introduced an LMS based equalizer [4], which does not operate with complex coefficients, but with real ones. This is unusual but makes it possible to handle I-branch and Q-branch imperfections independently. The I-branch and Q-

branch filter imperfections are generated by the analog base-band filters, which are two real filters. The IQ amplitude error detection will be done via equation (6).

$$e_I[n] = I[n] - \tilde{I}[n] \qquad (6)$$
$$e_Q[n] = Q[n] - \tilde{Q}[n]$$

From the ideal transmitted symbols there will be subtracted the imperfect estimated symbols. The calculated errors of both branches need not to be the same values. Hence there have to be calculated for both branches independent correction coefficients. This is described by equation (7). The new coefficients at the time n+1 will be calculated from the current coefficients at the time n and an additional addend.

$$\underline{c}_{I,Q}[n+1] = \underline{c}_{I,Q}[n] + \mu\, e_{I,Q}[n]\underline{D}_{I,Q}[n]\underline{h}_{I,Q}^{\#}[n] \qquad (7)$$

The addend consists out of four factors. First the constant μ describes the step width. The step width defines the loop accuracy, loop adaptation speed or loop bandwidth. Because the expected IQ amplitude imbalance errors will not change over a very long period of time the loop bandwidth needs not to be large and hence the loop accuracy can be high. The second factor is the calculated error from equation (6). After that the product of the ideal input data matrix $\underline{D}$ and an approximation $\underline{h}^{\#}$ [3] of the analog filters $\underline{h}_{I,Q}$ follows. The approximations of both analog filters will be simple tap-delay lines, providing the same latency as the analog filters contain. After the update of the coefficient vectors the correction takes place in programmable FIR filters.

VII. HW and SW Partitioning

To enable a flexible radio design it is useful to implement some functionality via software and not only hardware. The advantage is that software code can be optimised in an existing radio environment. A disadvantage might be that a Digital-Signal-Processor (DSP) has not enough processing power to calculate high data rate functions in time.

Both new adjustment loops require three different steps. The first one takes care about the IQ sample estimation from the envelope. The second provides the error detection of the IQ phase and IQ amplitude imperfections. The last step handles the error corrections. Assuming that the IQ imperfections are stable over a long period of time and they will not change their values rapidly it is possible to employ for the IQ sample estimation and the IQ error detection algorithms a DSP-based software approach. This is possible because the low rate of error value changes does require only a low rate of error detection updates

and a low rate of the IQ sample estimations. Hence a DSP, which might not handle the same operations on symbol rate in time, will be able to process now a limited number of operations based on a lower sample rate.

Assuming that the DSP processing power is not enough to handle the symbol rate based error corrections it is still necessary to employ hardware for both IQ correction blocks. Figure 4) provides a hardware-software partitioning overview. Three blocks from figure 1) have been replaced by the DSP. In this new architecture the envelope measurement results and ideal IQ samples will be transferred through the data bus to the processor. The instruction memory provides the software-based algorithms and the correction values will be transferred by the data bus to the IQ correction blocks.

The advantage of the software-based approach is the chance to change the IQ sample estimation and error detection after the overall system implementation has been finalized. The success of the IQ compensations is strongly depending on the algorithm development based on the modelling quality of the analog components. Might the analog components change with regards to the used simulation models, it is an advantage to be able to change the IQ sample estimation and IQ error detection, too. Changes of the analog components might appear because of technical or commercial reasons.
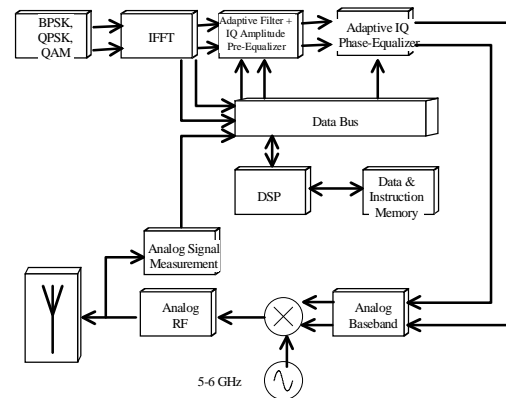

Fig. 4) DSP-based IQ estimation and error detection.

VIII. Simulation Results

Figures 5)-8) show different IQ diagrams in an IEEE802.11a OFDM environment.

Figure 5) provides the imperfect IQ symbols after ideal down-modulation. The imperfections were caused by IQ phase and IQ amplitude errors. Frequency selectivity has been added by imperfect analog base-band filters. Both base-band filters have slightly different transfer-functions.
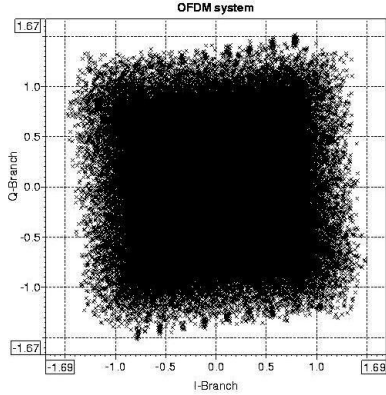
Fig. 5) Imperfect IQ diagram. $\varphi = -10°$. $a = 0.7, b = 1.0$.

Figure 6) provides the first corrected results. A 19-coefficient filter pre-equalizer and a non-frequency selective IQ phase adjustment do not provide full IQ compensation.
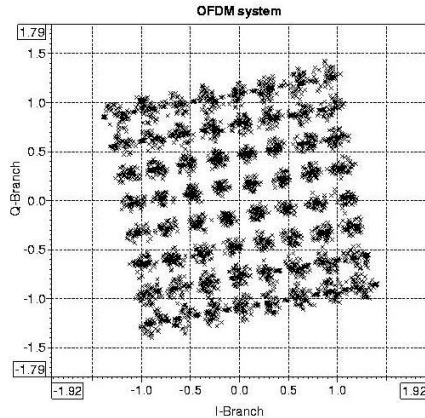

Fig. 6) 1-coefficient IQ phase adj., 19-coefficient IQ amplitude adjustment. $\varphi = -10°$. $a = 0.7$, $b = 1.0$.

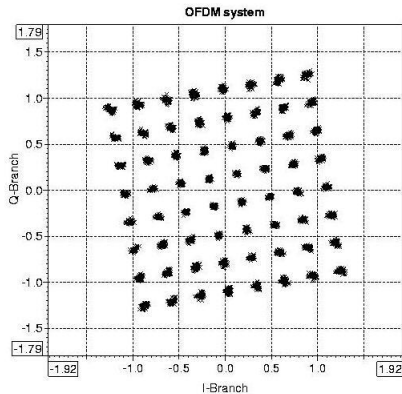Figure 7) differs from figure 6) only because the IQ phase pre-equalizer coefficient number equals 3.

Fig. 7) 3-coefficient IQ phase adj., 19-coefficient IQ amplitude adjustment. $\varphi = -10°$. $a = 0.7$, $b = 1.0$.

Figure 8) provides the required accuracy. Both adjustment loops employ 19-coefficients for each correction loop. The constellation points provide the required accuracy.
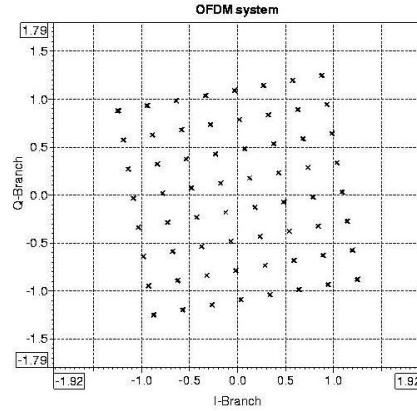

Fig. 8) 19-coefficient IQ phase adj., 19-coefficient IQ amplitude adjustment. $\varphi = -10°$. $a = 0.7$, $b = 1.0$.

The remaining phase shift of about -10° has been generated by the IQ phase error and will not be compensated.

IX. Conclusion

This paper provides two different time-domain algorithms to handle frequency selective IQ phase and IQ amplitude imbalance errors in an OFDM transmitter. By partitioning the algorithms into hardware and software it is possible to adapt critical parts of the digital algorithm based on new realizations of the analog components. In the case of a software-based approach it is important to consider the loop update rate. The simulation results show that the algorithms will provide significant improvements to the IQ samples in case of frequency selective IQ imperfections.

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