Design and Implementation of Self-Calibration for Digital Predistortion of Power Amplifiers

Dua Idris^{1,2}, Yannick Le Moullec^{1,2}, Patrick Eggers¹ ¹Center for Software Defined Radio ²Department of Electronic Systems Aalborg University Niels Jernes Vej 12, DK-9220 Aalborg East, Denmark

Abstract: The linearization of Power Amplifiers (PAs) on mobile handsets is a critical problem for stringent systems such as Wimax. In this paper we present the results of our investigation on the feasibility of self-calibration from a theoretical point of view as well as on a FPGA-based hardware platform for the predistortion of PAs. We propose a system design in three phases i) self-calibration, ii) inverse function calculation and iii) predistortion. MATLAB simulation results show that the functionality of the self-calibration, inverse function calculation and predistortion can linearize the PA charateristics and that the FPGA implementation results of the self-calibration are promising.

Key-Words: DPD, self-calibration, PA

1 Introduction

The distortion caused by power amplifiers can be classified as linear and non-linear [1]. The most significant impairment is caused due to the non-linear behavior of the amplifier. In standards such as WiMAX that are based on OFDM, another major problem of concern is the Peak to Average Power Ratio (PAPR))[2]. Superposition of the subcarriers of an OFDM signal results in strong variations of the instantaneous signal power i.e. high signal peaks which are ultimately distorted non-linearly by the power amplifier. Non-linear distortion results in the formation of intermodulation products. Inter Carrier Interference (ICI) is caused by intermodulation products that disturb the transmitted signal whereas Out-Of-Band (OOB) radiation is due to intermodulation products located outside the transmission band which disturb signals transmitted on adjacent frequency bands. There are several techniques used for reducing the PAPR of the signal such as clipping and filtering, coding etc. Another problem with WiMAX standard is the use of non-constant envelope modulation schemes that places a strict requirement on the power amplifiers to operate linearly.

2 Predistortion

In order to achieve a good spectral efficiency, higher order modulation schemes are used such as QAM which require well designed pulse shaping to reduce the out-of-band emissions. This kind of filtering places a requirement of a linear transmit amplifier due to the creation of a non-constant envelope signal that can provide fidelity to the varying signal envelope. Even if each signal has a constant envelope when several such signals share a wideband amplifier the composite signal will have a non-constant envelope. Class A amplifiers can achieve this level of linear fidelity but with certain backoff but this reduces the power efficiency. Then there are other classes of amplifiers such as AB, B and C which have a good power efficiency but they have a non-linear response which is not desired since it creates broadening of the transmitted signal spectrum. Predistortion can be used to compensate for such a spectral spill over where in a signal is predistorted before power amplification in such a way that after amplifier distortion the resulting signal approximates the ideal signal. The basic idea is to predistort the signal before amplification in such a way that when the signal undergoes the PA distortion, the resulting signal approximates the ideal signal. The concept involves the use a predistortion function which compensates for the nonlinear gain of the amplifier so that the cascaded functions produce a constant gain as shown in figure 1.

In this work, we propose a self-calibration approach of which the concept is that in a mobile device is it beneficial to be able to extract the changing characteristics of the PA in order to i) avoid the traditional measurements done during the production phase and



Figure 1: Predistortion concept

ii) to adapt to changing conditions.

3 PA Characterization

The characteristics of the PA are needed to be able to calculate the inverse characteristics used in the predistorter. Hence the performance of the predistorter depends on the accuracy with which the characteristics are obtained. The main characterizations of the power amplifier are the 1dB compression point, second and third order intercepts. The 1dB compression point refers to the output power level at which the amplifiers transfer characteristic deviates from that of an ideal linear characteristic by 1dB. This can be obtained using a single tone excitation. For the second and third order intercept points, two tone test is required. When the voltage of the input signal is increased, the second harmonic increases in proportion to the square of the input signal whereas the third harmonic increases in proportion to the cube of the input signal i.e. they are increasing at a rate greater than the fundamental component. But at one point the harmonic components are equal to the fundamental. The second order intercept point is the signal level at which the second harmonic is equal to the fundamental and the third order intercept point is the signal level where the third harmonic is equal to the fundamental component. For higher order non-linearities multitone tests are needed to measure the Adjacent Channel Power Ratio (ACPR) which gives the degree of the signal spreading into the adjacent channels. AM/AM characteristics give the non linear relationship between the input power and output power whereas the AM/PM characteristics give show the effect of the input signal on the phase of the output signal.

The AM/AM and AM/PM characteristics of the PA^1 were obtained using a single tone Continuous Wave (CW) with power sweep from -27 to 0 dBm but including the attenuation it amounts to -33dBm to -6dBm.

The measurements were done for supply voltages ranging from 3.3 to 5V to be able to observe the varia-



Figure 2: Gain Characteristic of the PA for different supply voltages

tions in characteristics due to this factor and these are shown in figure 2 and in figure 3 for gain and phase shift characteristics respectively. It was observed that as the supply voltage drops, the gain decreases and the phase shift increases.



Figure 3: Phase Characteristic of the PA for different supply voltages

4 Algorithm and System Design

4.1 Comparison of the LUT and Polynomial methods

The system operation can be classified into two main processes: i) Digital Predistortion (forward) and ii)

¹Confidential model reference

Self-Calibration (feedback). The input signals are digitally predistorted before being amplified such that the overall characteristic is linear. This process is called the digital predistortion phase which is performed as a continuous process. The other process is called the self-calibration phase. It is important for the predistorter to estimate the characteristic of the amplifier so it could implement its inverse characteristic. Since these characteristics are time varying, the coefficients for predistortion need to be corrected. This update requires an adaptation algorithm, for instance the LUT and polynomial methods.

The input signal undergoes a complex multiplication with the inverse functions calculated by the predistorter. It is complex since both the amplitude and phase of the signal are being modified accordingly. The requirement of a complex multiplication is common for both the algorithms. In the LUT the correction values are stored in a table and indexing is used to select the required values. The only additional computation in this phase is calculation of index. In the case of polynomial method, the coefficients are calculated and stored in a table from where they are accessed by the polynomial function which calculates the inverse functions. This step of calculation is an added complexity for polynomial approach that is avoided in the LUT approach. Though less memory is required to store complex polynomial coefficients, it may not be preferred due to higher number of computations involved.

Once the amplifier has been characterized along its entire dynamic range to be able to obtain its nonlinear behavior, the obtained characteristics are stored in a table from which the inverse is calculated. For the LUT approach this would require scaling of gain, subtractions for phase error and memory to store the values. Then the inverse of these values is taken for using in the predistorter table. But for a polynomial approach, algorithm such as least squares(LS)[3] or matrix inversion is needed to calculate a polynomial fit to the data to extract the polynomial coefficients that are to be stored in a table. Then inverse of these coefficients is calculated to be stored in the predistorter LUT. The calculation of polynomial fitting curve is computationally complex compared to the LUT approach. Since the characteristics vary with time, the predistorter correction values need to be updated. The output signal from the PA is downconverted and fedback to be compared with the input signal. The predistortion function calculated is then compared with the one that is being implemented and if found to differ, the values are updated.

Table 1 compares the computational complexity of both the algorithms. Consider M to be the number of multiplications and A to be the number of additions. The value of N is usually 32, 64, 128 or 256 where N is the number of samples, and K is the order of the polynomial which could be 3,5,7 or even higher.

Operation phase	LUT	Polynomial
DPD	(2M + 1A) and 1 CM	2(K-1)A, 2(2K)M + 1 CM
Self-calibration	N*M + N*A	(2N + 1)A and $(2N + 1)M$

Table 1: Computational complexity comparison between LUT and polynomial. M:Multiplication, A:Addition, CM:Complex Multiplication, N:Number of samples, K:order of the polynomial

Table 2 compares the memory requirements for both the algorithms, with N the number of samples and C the number of coefficients.

Operation phase	LUT	Polynomial
DPD	2Nm	2Cm
Self-calibration	3Nm	(2N + 2C)m

Table 2: Memory requirements comparison for LUT and Polynomial. m:memory units, N:Number of samples, C:number of Coefficients

Considering the above two criteria, the LUT approach has been chosen over the polynomial approach for implementing predistortion.

4.2 System Design

This is divided into three main procedures: Self calibration (measurement of PA characteristic), Inverse Function Calculation and Predistortion, as shown in figure 4 and figure 5.

4.2.1 Calibration

In order to characterize the PA, the calibration signal is sent through the PA and the output is downconverted, measured and stored. This process is done once during boot up or during idle time. Switch S1 and S2 are open and switch S2 is closed. The characterisation is done over the dynamic range of the PA. The steps can be chosen to uniform or non-uniform. In [4] it was shown that simple amplitude based uniform spacing is the most practical and near optimum choice to linearize the PA. Hence, equal step size is chosen and is calculated as per [5]: s = (Vmax - Vmin)/Nwhere s is the step size, Vmax is the maximum input voltage, Vmin is the minimum input voltage and N is the total number of table entries.

The PA has been characterized for 200 points with the calibration procedure shown in figure 4. Ii and Qi

represent the Inphase and Quadrature phase components of the calibration signal whereas Io and Qo represents the output from PA which is downconverted and fedback to the baseband processor through the ADCs. Attenuation is used to compensate for the gain of the PA before downconverting the signal. Polar format of representation has been preferred as it is easier to calculate the inverse. The samples of cartesian signals are used to calculate the magnitude and phase. The input magnitude and output magnitude are represented by ri and ro respectively whereas $\delta \tau$ represents the phase difference between the input and output samples. These values are stored in the table sequentially per sample.



Figure 4: Calibration phase of the power amplifier

4.2.2 Inverse Function Calculation

When the PA has been characterized, all the measured input and output magnitudes and the phase difference per sample are stored in the look up table. The next step is the calculate the inverse characteristic for the PA. This means that the gain and phase correction factors for the input samples have to be calculated. The gain correction factor compensates for the gain compression and the phase correction factor compensates for the phase shift per input sample. A simple algorithm is used for the inverse estimation and is explained in [5]. Since the AM/AM characteristic of the PA is nonlinear, it has to be linearised. In order to make the AM/AM characteristic a linear one, the magnitude has to be scaled by a factor. The calculation of this factor is explained with an example assuming the gain needed is one for simplicity. In order to compensate for the phase or AM/PM distortion, the negative of the phase-shift is added to the input signal phase. In the above example, the phase correction factor would be calculated as follows. At the 11th entry the phase shift is 0.1365 radians and if the phase of the input signal is shifted by -0.1365 radians, the phase would

not be distorted. This factor is stored in the inverse table for correcting the phase of the input.

4.2.3 Predistortion

The incoming sample is predistorted when it undergoes a complex multiplication with the corresponding values in the inverse look up table to obtain the inverse function so that the output of the PA can be linear. Figure 5 shows the implementation of digital predistortion. The input signal is converted to polar co-ordinate. The amplitude is used to index through the inverse look up table from which the correction factors are obtained and used to predistort the incoming signal. Ipd and Qpd represent the predistorted signal. The calibration process is done online. Followed by the inverse look up table calcu-lation which can be performed as post processing of the measured data. Then the table estimated from the calibration process is discarded and the inverse look up table is used to perform predistortion which can be done online.



Figure 5: Digital Predistortion in Polar Co-ordinate

4.3 MATLAB simulation results

The predistortion functionality is implemented in MATLAB to test the performance of the inverse function algorithm. The most relevant results are shown in the following figures.

Figure(3.7) shows the gain of the PA versus the input power. The compression of the gain at higher input power can be observed.

Figure(3.8) shows the inverse gain function which causes the gain expansion in order to compensate for the gain compression, such that the overall gain of the PA is linear.

Figure(3.12) compares the output power vs the input power for the case with and without predistortion and it is observed that after predistortion, the PA behaves linearly as expected.



Figure 6: Gain Vs Input Power for the PA without predistortion



Figure 7: Gain expansion vs Input Power



Figure 8: Output Power vs Input Power with and without predistortion

5 FPGA Implementation of the selfcalibration process

5.1 Setup and tools

In order to evaluate the feasibility of the proposed approach, the calibration phase (note that the FPGA implementation of the other phases, inversion and predistortion, are still under investigation) has been implemented on a XtremeDSP kit featuring a Virtex-4 FPGA and tested with the PA connected through the ADCs and DACs of the kit. The tool used to program the FPGA is Xilinx's System Generator, a high level tool used inside Simulink/Matlab, providing a convenient and rapid path to FPGA prototyping. Furthermore, we have used the co-simulation capabilities of System Generator to exchange data between Simulink and the FPGA.

5.2 Testing and Results

5.3 RF Setup

For testing the calibration process with PA in the loop, the RF setup made with off the shelf mini circuit components as shown in figure 9.

Figure 9: RF setup for testing the power amplifier

The I and Q signals from the DACs are low pass filtered using filters with a bandwidth of 30MHz. The filtered signals are fed to the quadrature mixer. The output is filtered through a bandpass filter with bandwidth of 3MHz. The combined signal is upconverted to 2.43GHz. A preamplifier is added to boost the signal level by 10dB. This is followed by a bandpass filter to remove the spurious components due to amplification. Finally, the signal is amplified by the PA that has a gain of 34dB. In order to feedback the output of the amplifier back to baseband, the amplified signal has to be downconverted. But before this, it has to be attenuated so that the downconverter can tolerate or handle the signal. An attenuator of 30dB is used before downconversion. The downconverted signal is bandpass filtered before giving it to a quadrature mixer. This mixer produces I and Q signals which are bandpass filtered and then given to the ADCs.

5.4 Results

5.4.1 PA operated in Linear Mode

Figure 10 shows the output magnitude vs input magnitude plot when the amplifier was driven in the linear region. The input power is restricted to be below -5dBm. Due to noise floor, the curves are almost flat in the lower range of input magnitude and begins to rise linearly after 0.05V but at the higher input magnitude range.



Figure 10: Output magnitude(in volts) vs input magnitude(in volts) for several supply voltages obtained from calibration procedure implemented on the FPGA when the PA is operating in the linear region

5.4.2 PA operating under Compression

Figure 11 shows the output magnitude vs input magnitude plot when the amplifier was driven into the compression region. The input power exceeds -5dBm which is specified to be the maximum input power for the PA. It can be observed that the PA begins to saturate after it crosses the maximum input power allowed i.e. -5dBm.

6 Conclusion and Future Works

6.1 Conclusion

The need for dynamically linearising PAs on mobile handsets is a critical problem for stringent systems such as Wimax. In this paper we have investigated the feasibility of self-calibration from a simulation point of view (Matlab) as well as on a FPGA-based hardware platform. Firstly two approaches have been compared (LUT vs. polynomial). Our results shows



Figure 11: Output magnitude(in volts) vs input magnitude(in volts) for different supply voltages obtained from calibration procedure implemented on the FPGA when the PA is operating under compression

that the LUT is cheaper than the polynomial method in terms of implementational complexity. Secondly, the system has been designed in three phases i) selfcalibration, ii) inverse function calculation and iii) predistortion. Thirdly, MATLAB simulation results shows that the functionality of the self-calibration, inverse function calculation and predistortion can linearize the PA characteristics. Finally, we designed, implemented and tested the self-calibration process on a FPGA and tested it using the PA. Future works include the FPGA implementation of the inverse function calculation and predistortion blocks, testing of the system performance with a WiMAX signal and making more considerations for implementing this technique on a real mobile handset (ASIC instead of FPGA).

References:

- [1] National Instruments, Sources of Error in I/Q Based RF Signal Generator.
- [2] A. Saul, Ed., Analysis of Peak Reduction in OFDM Systems Based on Recursive Clipping, Vol. 1. Proc in Int. OFDM Workshop, September 2003.
- [3] T. Lynge Kjeldsen et al., Adaptive Estimation of a Wireless Channel, Aalborg University, Applied Signal Processing and Implementation, 8th Semester Project Report, 2006.
- [4] J. K. Cavers, Optimum Table Spacing in Predistorting Amplifier Linearizers, in IEEE trans. on Vehicular Technology, vol. 48, no. 5, September 1999.
- [5] D.-S. Han and T. Hwang, An Adaptive Predistorter for the Compensation of HPA Nonlinearity, in IEEE transacions on broadcasting, vol. 46, no. 2, June 2000.