Master’s Thesis

Different Digital Predistortion Techniques for Power Amplifier Linearization

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Department of Electrical and Information Technology,
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Abstract

Linearity and high efficiency are crucial requirements for any power amplifier. However, power amplifiers have high efficiency levels when they are in non-linear regions. To overcome this issue there have been many suggestions in literature, one of the most successful methods is digital predistortion method.

Digital predistortion (DPD) method's low resource usage and fairly easy algorithm draws a lot of researcher's attention. Many different methods are suggested for DPD algorithms. Volterra series based methods draws even more attention due to its flexibility and easy implementation.

However, deciding which method to use for DPD purposes is not completed even when Volterra series based method is chosen. There are many different Volterra series based methods which differ from each other. This paper examines 5 different Volterra based methods for DPD purposes and tests them in 2 different PAs with LTE signals. Also forward behavioral modelling performance of these 5 methods are also examined for each PA with same signals.

In chapter 1 needed theoretical information is explained about power amplifier characteristics. In chapter 2, the five chosen methods are examined in detail and corresponding parameters are explained. In chapter 3, forward behavioral modelling setup and the way to model power amplifiers is explained. In chapter 4, setup and the digital predistortion algorithms are explained. In the 5th chapter the results of both behavioral modelling and DPD is shown and comments on the results are given.

Acknowledgments

I think this is the hardest part of my thesis, expressing my gratitude to the people who guided and helped me during my thesis, I don't think I can find enough words to express my thankfulness. Nevertheless, I will try my best!

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My fiancée, Necla Coskun, who supported me every single time when I had a problem not only with my thesis with my life she was there to support me, should be considered as a second writer for this thesis for her support until the mornings. I love you.

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This project has been done in Ericsson, Lund.

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1 Introduction

1.1 Wireless Link & Power Amplifiers

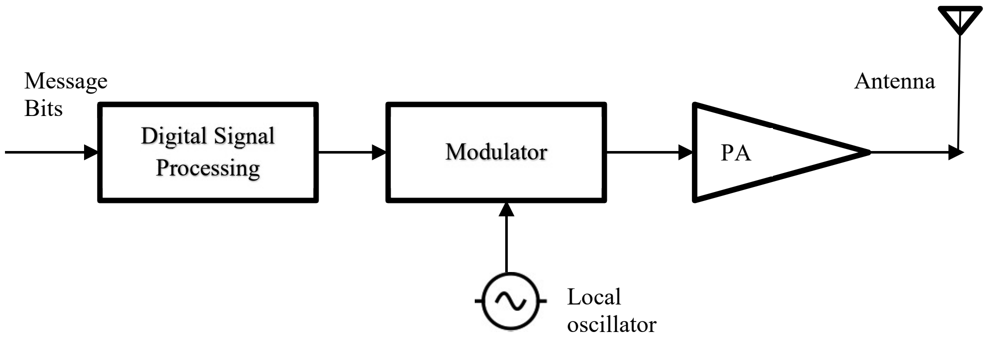

Figure 1-1: Block diagram of a simplified modern transmitter architecture

Power amplifiers (PA) are one of the indispensable block in wireless communication systems [1], which are responsible of overcoming the loss between the transmitter and receiver. Figure 1-1 shows a simplified block diagram of modern homodyne transmitter architecture. To achieve the sufficient transmission power, PAs are highly important in communication systems. By increasing the input power to the PA, high efficiency levels can be achieved. However, PAs are inherently non-linear when they are driven in high power modes, which creates an inevitable trade-off between efficiency and linearity. Moreover, latest techniques which are widely used in wireless communication industry due to their high speed of communication, that are controlled by 3GPP [2] such as Orthogonal Frequency Division Multiplexing (OFDM) and Wideband Code Division Multiple Access(W-CDMA) is using non-constant envelope techniques. While increasing the frequency spectrum efficiency, these techniques have high peak to average power ratio (PAPR). For the PA to be linear when transmitting the symbols with highest peak power, the average PA output power need to be several dB lower than the compression point approximately PAPR dB lower, which decreases the overall efficiency [3]. To overcome this non-linearity - high efficiency problem one of the widely accepted techniques is digital predistortion (DPD) [4], [5], [6]. In this paper different DPD techniques will be examined and implemented.

1.2 Power Amplifier Efficiency

Energy saving technologies have been developed in all areas of electronics mainly because of the increased CO2 emissions which causes global warming. Designers are working on decreasing the CO2 emissions of power plants and also increasing the power efficiency of the systems. This creates pressure of increasing the power efficiency of modern wireless communication system to its designer. Among all the Radio Frequency (RF) components, power amplifiers (PAs) have the highest consumption of power which can be up to 40 percent of the overall power budget [7]. “A fully-loaded 3G cell utilizing legacy power amplifier technology may consume about 3 kW of power, which gives a cost of 1600 US dollar of electricity per year in US, or 2300 Euro in Europe. A European operator with a network of 20 000 base stations would draw about 60 MW, giving an electricity cost per year of 46 million of Euro, and a carbon footprint around 220 000 tons of CO2 per year” [7]. These statistics that are mentioned in [7] by Yarleque, indicates the importance of the power efficiency of a PA in a base station. PA’s efficiency increases with the input power, figure 1-2 below shows this relation for one of the PAs that is used in this project. The technique that is examined in this paper focuses on increasing the power efficiency without distorting the signal properties and without violating the frequency spectrum regulations.

The method to calculate the efficiency of a PA is called Power Added Efficiency (PAE) which can be described as:

$$PAE = \frac{P_{RFout} - P_{in}}{P_{DC}}$$

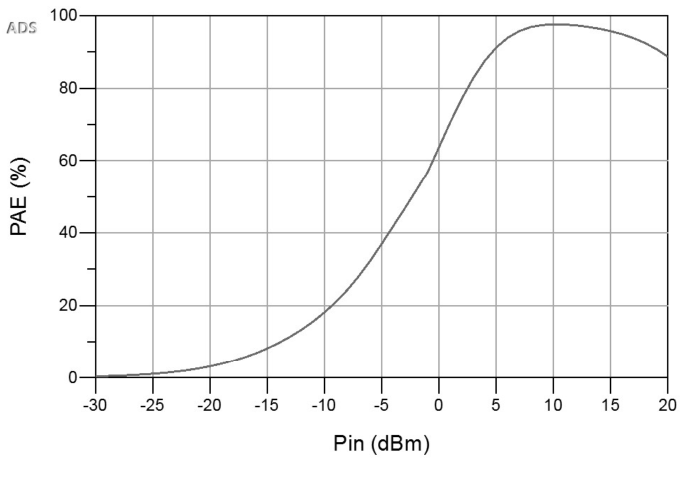

Figure 1-2: Input Power vs PAE

1.3 Importance of Linearity & Non-Constant Envelope Transmission

Latest trending communication techniques both used in 3G (W-CDMA) and 4G (OFDM) uses non-constant envelope schemes. In figure 1-3 a sample 256 QAM (which uses 256 different constellations points) OFDM signal is shown. As mentioned before these techniques also has high PAPR which creates the conflict between power efficiency and frequency spectrum efficiency (linearity). Until so far in the paper, the reasons of this conflict and the importance of the power efficiency is explained. Now the importance of the linearity will be explained. When a PA is non-linear it starts to create harmonic distortion products which causes frequency spectral growth. However, this needs to be avoided due to strict frequency allocations.

In Europe the frequency spectrum is controlled by European Conference of Postal and Telecommunications Administrations (CEPT). The details about the allocations can be found in [8]. These allocations cannot be violated for any wireless communication technique for any carrier frequency.

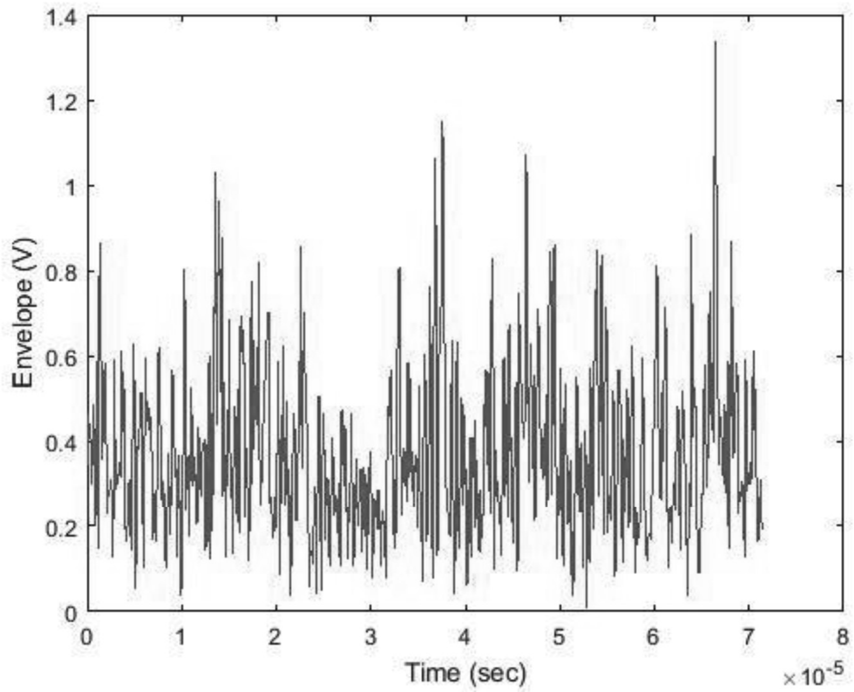

Figure 1-3: Sample 256 QAM, OFDM signal

3GPP standards, define the maximum acceptable Adjacent Channel Power Ratio (ACPR) for the mobile phone communications for data transmission for 3G, LTE (4G). The details about these regulations can be found in [2]. However, in this report the ACPR level that is checked is called ACPR1 and it can be seen on figure 1-4. ACPR1 is the neighboring channel, there are also other requirements about further out channels, however they are not considered in this project.

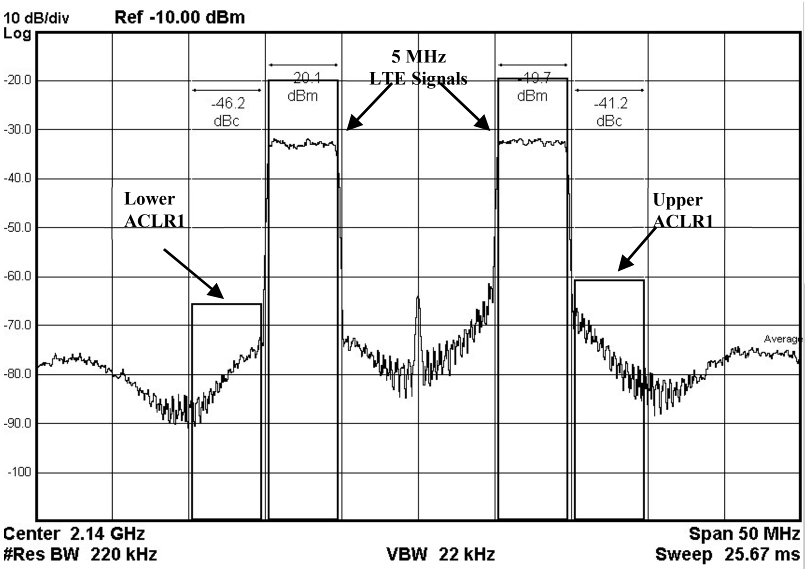

Figure 1-4: ACPR1 regions for 2 cascaded 5 MHz LTE signals with 15 MHz spacing

1.4 Non Linearity Effect of Power Amplifiers

So far the importance of linearity is emphasized yet the reasons of PA's non-linearity is not discussed. In this section the reasons of PA's non-linearity will be discussed. To achieve high power efficiency levels, a PA should be driven in high power mode. After a certain input power each PA enters the compression region. A power amplifier has constant gain for low level input signals, however for high power levels PA goes in to saturation and the gain is decreased, the point where the gain is decreased 1 dB from it's constant value is called 1 dB compression point which can be seen in figure 1-5 [9]. When the PA is in compression region the output signal's peak points becomes clipped.

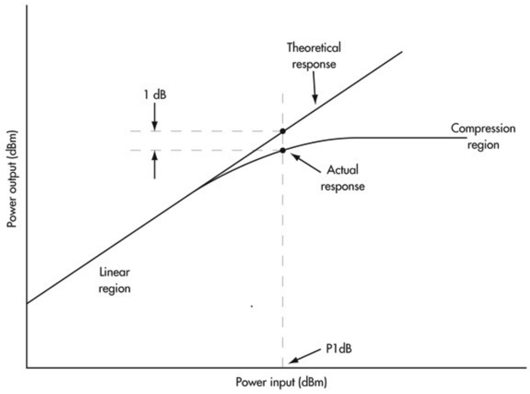

Figure 1-5: Input power vs output power for 1 dB compression point

This clipping in time domain has a spectral growth effect in frequency domain. Figure 1-6 and figure 1-7 are showing the effect of clipping, in figure 1-6 a sinusoidal wave of 1 MHz frequency and it's frequency domain response is shown. In figure 1-7 same wave but with five percent clipping and its frequency response is shown. As it can be seen, when there is clipping, frequency spectrum of the signal grows even for a single tone input signal. Moreover, if the results of two tone test is examined figure 1-8 from [10], one can see the intermodulation products in frequency domain caused by non-linearities in power amplifiers when the PA is in compression region. In wireless communication, intermodulation products that are adjacent to signal are more important, rest of the harmonics and intermodulation products can be suppressed by using filters.

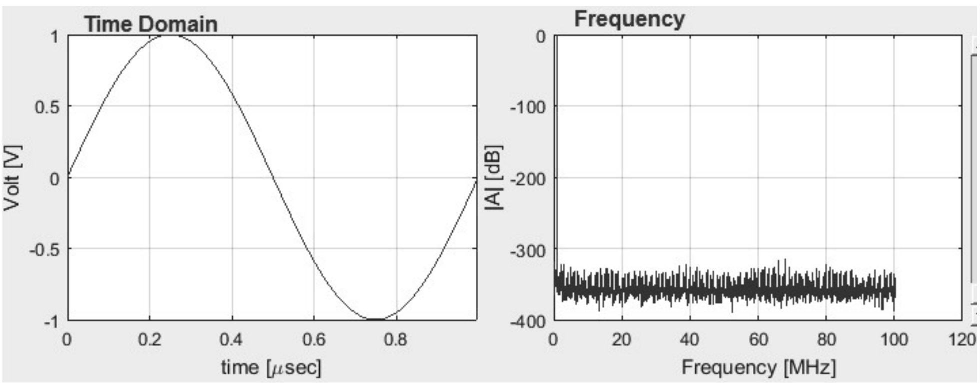

Figure 1-6: 1 MHz Sinusoidal wave without clipping & it's frequency response

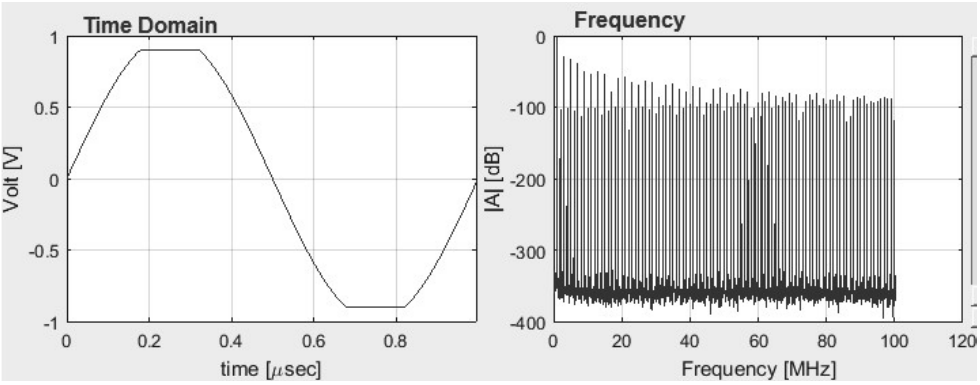

Figure 1-7: 1 MHz Sinusoidal wave with clipping & it's frequency response

In real scenario there are infinite tones in a transmitted signal, for example in LTE, signals have up to 20 MHz bandwidth and in later version of LTE signals will have 100 MHz bandwidth. For large bandwidths the frequency spectrum is ruined if the PA is in non-linear region, a sample case is shown in figure 1-9 for 20 MHz LTE signal. These non-linear characteristics of a power amplifier makes linearization procedures inevitable for high power efficiency modes.

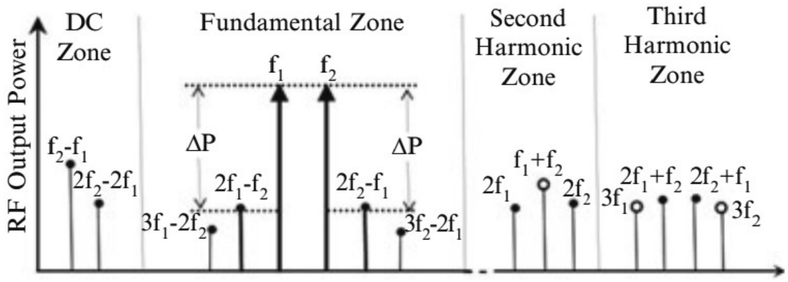

Figure 1-8: 2 tones test results of a PA, when the PA is in non-linear region [10]

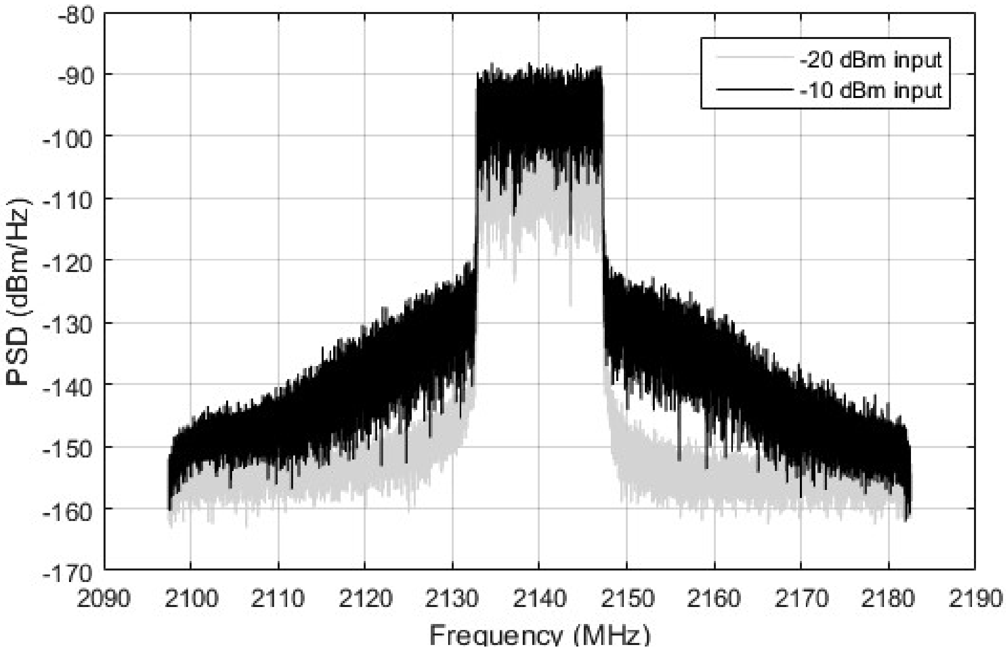

Figure 1-9: Comparison of 2 different input signals with different power levels

1.5 AM/AM & AM/PM Distortions

A power amplifiers non-linear effects can be examined by amplitude to amplitude (AM/AM) conversion and amplitude to phase (AM/PM) conversion. AM/AM can be described as the deviation from the constant gain when the input power is increased towards compression region and AM/PM can be described as the change of the phase of signal caused by PA when the input power is increased towards compression region.

Amplitude to Amplitude distortion occurs when the PA enters the compression zone. In figure 1-5 the 1 dB compression point is shown, it is the characterization of the PA for different input power levels. If the input signal has high PAPR values, the peaks of the input signal will drive the PA in to compression region and the gain of the PA will deviate from its constant value. Instead of leaving a back off region as mentioned before, a DPD algorithm can be used to fix this distorted AM/AM conversion in compression region.

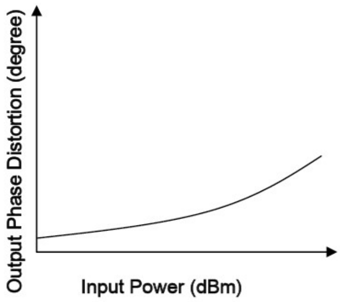

Figure 1-10 AM/PM Conversion

Moreover, when a PA enters the compression region the change of the phase of the signal caused by PA starts to increase, figure 1-10 from [11] shows this increase. A DPD algorithm also can fix this problem and decrease the phase effect of the PA.

The effects of the DPD to AM/AM and AM/PM distortions of the PA will be discussed in the results section. In figure 1-11 one can see the AM/AM and AM/PM curves of a PA, which belongs to one of the PAs that is used for this projects.

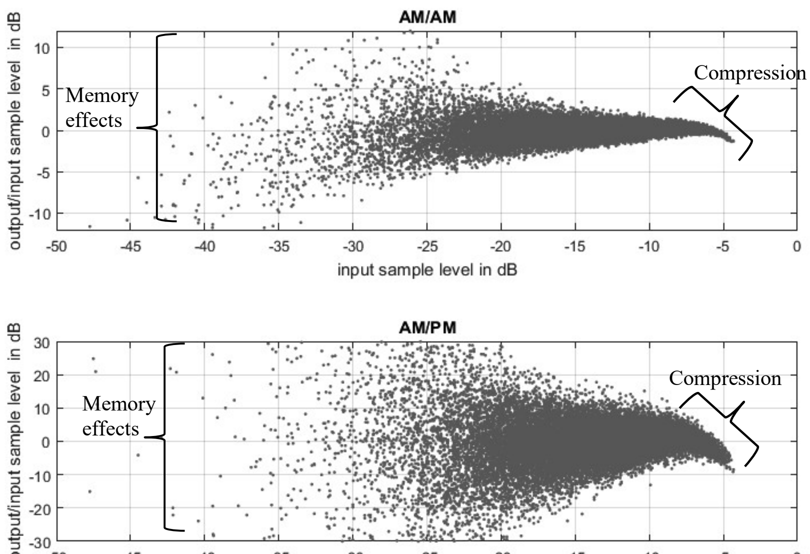

Figure 1-11: AM/AM and AM/PM curves

1.6 Thesis Outline

The thesis is organized as follows. In chapter 2, after discussing Volterra series, 5 different Volterra Series based modelling methods are discussed. Mathematical properties of these methods and the complexity in terms of coefficient numbers is discussed.

In chapter 3 behavioral modelling of PA by using 5 different methods is discussed in detail, simulation methods are explained and the coefficient estimation algorithm of each method is discussed.

In chapter 4 by using 5 methods that are explained in chapter 2, the digital predistortion algorithms are explained and also the measurement setup is shown.

In chapter 5 simulation results of chapter 3 and experiment results of chapter 4 are compared and discussed. After this chapter, conclusions and future work suggestions are discussed.

2 Volterra Series

2.1 Volterra Series Background

Volterra series [12] is a mathematical series that can represent non-linear systems which has memory effects. The continuous time Volterra series can be described by:

$$\boldsymbol{y(t) = h_0 + \sum_{n=1}^{N} \int_a^b \dots \int_a^b h_n(\tau_1, \dots, \tau_n) \prod_{j=1}^{n} x(t-\tau_j)\, d\tau_j} \qquad 2\text{-}1$$

Where x is the input, y is the output, and $h_n(\tau_1, \tau_2)$ is the kernels of the system to be modeled. Kernels are the coefficients which defines each system to be modeled. After estimating the kernels, a system can be modeled for any input value. If a system's output only depends on the current and previous input values, it is considered as a casual system. Power amplifiers can be considered as casual systems. In this case integral region starts from 0 or higher. Moreover, in the absence of input if the output is zero, h_0 in the formula becomes zero – which is the case for power amplifiers.

In digital systems, discrete time Volterra series [13] is needed and it can be described by:

$$y(k) = \sum_{n=0}^{\infty} y^{(n)}(k)$$

2-2

$$y^{(n)}(k) = \sum_{i_1=-\infty}^{\infty} \sum_{i_2=-\infty}^{\infty} \dots \sum_{i_n=-\infty}^{\infty} h^{(n)}(i_1, i_2, \dots, i_3) x(k-i_1) x(k-i_2) \dots x(k-i_n)$$

2-3

Where, $h^{(n)}$ is the kernels of the system, x is the input and y is the output. If the discrete formula is examined, it can be understood that it models any system by using impulse responses of the system and in this modelling i is the memory instance and n is the power order. Partial responses [13], i.e. $y^{(n)}(k)$ until the second power order can be written as:

$$\boldsymbol{y^{(0)}(k) = h^{(0)}(k)}$$

2-4

$$\boldsymbol{y^{(1)}(k)} = \sum_{i=-\infty}^{\infty} \boldsymbol{h^{(1)}(k, i) x(i)}$$

2-5

$$\boldsymbol{y^{(2)}(k)} = \sum_{i_1=-\infty}^{\infty} \sum_{i_2=-\infty}^{\infty} \boldsymbol{h^{(2)}(k, i_1, i_2) x(i_1) x(i_2)}$$

2-6

The general discrete Volterra series in 2.2 cannot be used in hardware applications due to high complexity and high need of resources. Equations 2-4,2-5,2-6 are discrete Volterra series without any truncation until second power order, if these equations are examined one can see that it is not possible to use the discrete Volterra series in PA modelling due to it's complexity. A truncation should be made on the Volterra series. However, this truncation should be made in a way that it should affect the performance of the modelling as little as possible. This can be done by omitting some of the terms in 2.2 for each specific application. To be able to use Volterra series in PA behavioral modelling, special versions of 2.2 is published. In this paper 5 different versions of these specialized Volterra series is examined both for behavioral modelling of a PA and digital predistortion algorithms. Details about these methods can be found at the next section.

2.2 Different Volterra Based Models

As it was mentioned in the previous section Volterra series is capable of modelling any non-linear system which has memory effects [14]. However, in real time applications truncation of some specific terms is needed. There have been many articles which suggests different truncations, in this project following 5 methods are chosen to be examined and tested. These following methods can be used to model any system which is non-linear and has memory effects, however in this paper these methods are only tested for power amplifiers.

2.2.1 Memory Polynomial

Memory polynomial [15] is a special case of Volterra Series which does not have all of the terms in Volterra series. To model the memory effect and non-linarities of a power amplifier in a digital system, truncations have to be done to original Volterra series due to the limited resources and limited estimation time. One of these truncated models called memory polynomial (MP) and can be formulated as:

$$y_{MP}(n) = \sum_{k=1}^{K}\sum_{m=0}^{M} a_{km}x(n-m)|x(n-m)|^{k-1} \tag{2-7}$$

Where x is the input, y is the output, K is the maximum power order, M is the maximum memory depth, and a_{kq} is the kernels (coefficients) of the system.

As it can be seen from 2.7 MP is a method that models the system by only considering the input signal and input signal's envelope for the same memory instance. It can also be thought as; MP method only considers the diagonal terms of the Volterra series. By changing K and M one can have control over the memory length and the non-linearity order of the system.

2.2.2 Generalized Memory Polynomial

To improve the performance of the MP method, many different techniques has been suggested. Generalized memory polynomial (GMP) [16] is one of these techniques. The difference between GMP and MP is that, in GMP there are extra terms to compare the terms in MP. In other words, by adding some extra terms to MP in 2.7 performance of the modelling of the system can be improved. The GMP technique can be described as:

$$y_{GMP}(n) = \sum_{k=1}^{K_a}\sum_{m=0}^{M_a} a_{km}x(n-m)|x(n-m)|^{k-1} + \sum_{k=1}^{K_b}\sum_{m=0}^{M_b}\sum_{l=1}^{L_b} b_{kml}x(n-l)|x(n-m-l)|^{k} + \sum_{k=1}^{K_c}\sum_{m=0}^{M_c}\sum_{l=1}^{L_c} c_{kml}x(n-l)|x(n-m+l)|^{k} \qquad \textbf{2-8}$$

Where K_a and M_a are the non-linearity order and memory depth of the diagonal terms. K_b, M_b, K_c, M_c are the non-linearity order and memory depth of the cross terms and L_b, L_c are the values that controls the leading and lagging terms in the method. Second line of 2-8 is a different version of first line of 2-8 by lagging the envelope compare to the signal itself, and the third line is same except instead of lagging the envelope is shifted forward in time (leading). The coefficients of the system are a, b, and c. In [16] it is shown that for some specific PAs this offered method – adding leading and lagging cross terms- can improve the modelling performance. By adjusting the parameters of the arrays one can change the performance of the modelling for each PA. In the next part another method which has additional terms to GMP will be explained.

2.2.3 Simplified Volterra

Research to improve the performance of the modelling of a power amplifier is never stopped. There have been many different suggestions about improving the performance as mention before. Simplified Volterra [17] is another method which is suggested to improve the performance. GMP method which has been mentioned in 2.2.2 is extended version of MP method, Simplified Volterra (SV) is the extended version of GMP method. Simplified Volterra can be described as:

$$
\begin{aligned}
y_{SV}(n) = & \sum_{k=1}^{K_a}\sum_{m=0}^{M_a} a_{km}x(n-m)|x(n-m)|^{k-1} \\
& + \sum_{k=1}^{K_b}\sum_{m=0}^{M_b}\sum_{l=1}^{L_b} b_{kml}x(n-l)|x(n-m-l)|^{k} \\
& + \sum_{k=1}^{K_c}\sum_{m=0}^{M_c}\sum_{l=1}^{L_c} c_{kml}x(n-l)|x(n-m-l)|^{k} \\
& + \sum_{k=1}^{K_d}\sum_{m=0}^{M_d}\sum_{l=1}^{L_d} d_{kml}x^{*}(n-l)x^{2}(n-m-l)|x(n-m-l)|^{k} \\
& + \sum_{k=1}^{K_e}\sum_{m=0}^{M_e}\sum_{l=1}^{L_e} e_{kml}x^{*}(n-l)x^{2}(n-m+l)|x(n-m+l)|^{k}
\end{aligned}
\tag{2-9}
$$

Where, K_a, K_b, K_c, K_d, K_e are the non-linearity orders of the arrays, respectively. M_a, M_b, M_c, M_d, M_e are the memory depth of the arrays, respectively. L_b, L_c, L_d, L_e are the leading or lagging cross terms indexes respectively. $a_{km}, b_{kml}, c_{kml}, d_{kml}, e_{kml}$ are the coefficients of the system. All of these K, M and L values are the parameters of the system which controls the different orders of the method. One has control over them individually, by adjusting these parameters one can change the behavioral modelling performance of the system for each PA.

In 2.2.1, 2.2.2 and 2.2.3; 3 different methods have been mentioned. If these methods are examined, one can realize that they have some similarities. One of these similarities is, all 3 of them have linear coefficients which means that by any least square estimation method these coefficients can be estimated. Details about coefficient estimation will be explained in the following chapter. Another similarity between these 3 methods is that they all use the input envelope and the input signal, yet by changing the time instances of these envelope and the signal multiplications; and also by adding extra terms to the arrays, the 3 methods differ from each other. Also the number of the coefficients are different from each other, increasing the complexity of the method means increasing the number of the coefficients. This is a tradeoff between performance and the resource which should be decided by the designer. By adjusting the values of parameters the coefficients can be controlled.

Methods in 2.2.4 and 2.2.5 are different from the previous methods in terms of the linearity of the coefficients. These two methods are the different approaches to 2.2.2 and 2.2.3 by having augmented coefficients.

2.2.4 Augmented Complexity-Reduced Generalized Memory Polynomial

The number of coefficients are as important as the performance of the method if the system has limited sources. This, encouraged researchers to develop new methods which decreases the number of coefficients without reducing the modelling performance of the system. Augmented complexity-reduced generalized memory polynomial (ACR-GMP) [18] is one of these methods. In this project main focus is on the performance of each method, complexity of each method is not considered. However, two different methods which focus on decreasing the complexity is examined to examine if they improve the performance or not.

In this method first the complexity of the conventional GMP method is decreased by splitting memoryless non-linearity and the memory effect of GMP. This operation resulted in decreased performance, to overcome this problem a parallel non-linear memory effect (NME) block is added. After this step the complexity is reduced, however a comparable performance modelling method is achieved. Details can be found at [18]. ACR-GMP method can be described as:

$$y_{ACR-GMP}(n) = y_{CR-GMP}(n) + y_{NME}(n)$$

2-10

$$y_{CR-GMP}(n) = \sum_{k=1}^{K_a} r_k^{(a)} \sum_{m=0}^{M_a} a_m\, x(n-m)|x(n-m)|^k + \sum_{k=1}^{K_b} r_k^{(b)} \sum_{m=0}^{M_b} \sum_{l=1}^{L_b} b_{ml} x(n-m)\, |x(n-m-l)|^k$$

$$+\sum_{k=1}^{K_c} r_k^{(c)} \sum_{m=0}^{M_c} \sum_{l=1}^{L_c} b_{ml} x(n-m)\,|x(n-m+l)|^k$$

2-11

$$y_{NME}(n) = \sum_{m=0}^{M_d} d_m |u(n-m)|^2 u(n-m)$$

2-12

$$u(n) = \sum_{k=1}^{K_a} r_k^{(a)} x(n)|x(n)|^{k-1}$$

2-13

K_a, K_b, K_c are the non-linearity orders of the diagonal, lagging and leading terms respectively. M_a, M_b, M_c are the memory depth of the diagonal, lagging and leading terms respectively. L_a, L_b are the parameters for controlling the memory depth of the lagging and leading terms of cross terms respectively. M_d controls the memory depth of the NME sub-block. In this method the first 3 terms are the different approaches of the conventional GMP method. As it can be seen, if the coefficients are disregarded they are the same equations with conventional GMP. Leading and lagging envelope terms added to MP method. In this method first memoryless non-linearity is considered for conventional GMP method, and after this, by using finite impulse response (FIR) filters memory effect is also added to each array. Lastly to improve the performance as mentioned before a NME parallel sub-block is added. This structure of arrays requires a new approach for estimating the coefficients. Estimation process cannot be completed in single loop; multiple loops are needed.

2.2.5 Augmented Complexity-Reduced Simplified Volterra

With the same approach as in section 2.2.4, Augmented Complexity Reduced Simplified Volterra method is established [19]. The difference with the 2.2.4 is that this method based on SV method and works with SV method's principals.

Similar to 2.2.4, also this method first separates the non-linearity and memory effect and combine them afterwards. These is an added NME sub-block to compensate the effect of this separation. ACR-SV method can be described as:

$$y_{ACR-SV}(n) = y_{CR-SV}(n) + y_{NME}(n)$$

2-14

$$y_{CR-SV}(n) = \sum_{k=1}^{K_a} r_k^{(a)} \sum_{m=0}^{M_a} a_m\, x(n-m)|x(n-m)|^k$$
$$+\sum_{k=1}^{K_b} r_k^{(b)} \sum_{m=0}^{M_b} \sum_{l=1}^{L_b} b_{ml} x(n-m)\, |x(n-m-l)|^k$$
$$+\sum_{k=1}^{K_c} r_k^{(c)} \sum_{m=0}^{M_c} \sum_{l=1}^{L_c} c_{ml} x(n-m)\, |x(n-m+l)|^k$$
$$+\sum_{k=1}^{K_d} r_k^{(d)} \sum_{m=0}^{M_d} \sum_{l=1}^{L_d} d_{ml}\, x^*(n-l) x^2(n-m-l)|x(n-m-l)|^k$$
$$+\sum_{k=1}^{K_e} r_k^{(c)} \sum_{m=0}^{M_e} \sum_{l=1}^{L_e} e_{ml}\, x^*(n-l) x^2(n-m+l)|x(n-m+l)|^k$$

2-15

$$y_{NME}(n) = \sum_{m=0}^{M_f} f_m |u(n-m)|^2 u(n-m)$$

2-16

$$u(n) = \textstyle\sum_{k=1}^{K_a} r_k^{(a)} x(n)|x(n)|^{k-1}$$

2-17

The similarities between ACR-GMP and ACR-SV can be easily seen by examining their definition equations. In ACR-SV two more arrays are added to ACR-GMP method which comes from the conventional SV method. ACR-GMP and ACR-SV are similar methods, to decrease the complexity -number of coefficients- first the memoryless non-linearities considered for conventional SV method and then by using FIR filters memory effect is also added. To compensate the reduce in the performance of modelling. K_a, K_b, K_c, K_d, K_e are the parameters that controls the non-linearity order of; diagonal terms, cross lagging and leading terms, complex conjugate lagging

and leading cross terms respectively. M_a, M_b, M_c, M_d, M_e are the parameters that controls the memory depth of; diagonal terms, cross lagging and leading terms, complex conjugate lagging and leading cross terms respectively. L_b, L_c, L_d, L_e are the parameters that controls the lagging and leading indexes of both cross terms and complex conjugate cross terms. M_F is the control of the NME sub-block's memory depth.
Same way with previous methods designer can control the number of coefficients by changing the parameters, it is a tradeoff between performance and the resource usage. The experiments that have been done for this thesis shows that increasing the number of coefficients will not always end up in better performance results in terms of modelling. Details will be explained in the following chapter.

2.2.6 Comparison of Number of Coefficients

The pursue of improving the performance of modelling a non-linear system resulted in new methods. These methods can be compared with each other in terms of performance or in terms complexity (resource usage-number of coefficients). In the upcoming parts each model will be compared according to their performances in behavioral modelling. From the digital predistortion perspective, complexity (number of coefficients) was out of scope of this project. The goal for this project is finding the best resulting method for linearizing the PA.

There are 2 important points that needs to be mentioned about the number of coefficients.
Firstly, designer has control over the parameters which means that designer can control the number of coefficients. The decision of the parameters; which controls the performance of the system and the number of the coefficients i.e. resource usage of the system, is a tradeoff which the designer has to decide. The simulations and also the test have been done for this paper showed that increasing the number of coefficients i.e. non-linearity order, memory depth, cross terms memory depths will not necessarily increase the performance. At this point, the designer should check and compare the system for different parameters. There are some methods to estimate the memory depth and non-linearity order in [20], [21], [22], however this is out of scope of this paper.
Secondly, each power amplifier can have different performances for the same parameter values, it is designer's responsibility to check the performance for different parameters, i.e. sweep the parameters and compare the performance of the power amplifier.
MP method's number of coefficients can be found by:

$$\boldsymbol{K(M+1)} \quad \textbf{2-18}$$

GMP method's number of coefficients can be found by:

$$K_a(M_a+1)+K_b(M_B+1)L_B \ +K_C(M_C+1)L_C$$

2-19

SV method's number of coefficients can be found by:

$$K_a(M_a+1)+K_b(M_B+1)L_B \ +K_C(M_C+1)L_C+K_d(M_d+1)L_d+ K_e(M_e+1)L_e$$

2-20

ACR GMP method's number of coefficients can be found by:

$$K_a+K_b+K_c+(M_a+1)+(M_b+1)L_b+(M_c+1)L_c+M_d+1$$

2-21

ACR SV method's number of coefficients can be found by:

$$K_a+K_b+K_c+K_d+K_e+(M_a+1)+(M_b+1)L_b+(M_c+1)L_c+(M_d+1)L_d+(M_e+1)L_e+M_f$$

2-22

3 Power Amplifier Behavioral Modelling

3.1 Volterra Series Based Modelling

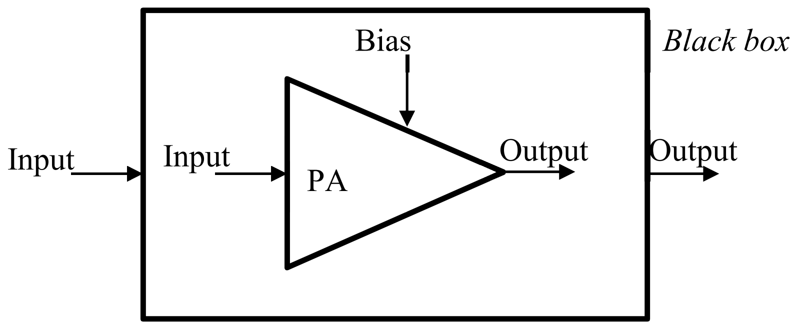

Figure 3-1 Basic Power Amplifier and blackbox diagram

In figure 3-1 a general power amplifier is represented. Behavioral modelling allows a designer to simulate and understand behavior of an amplifier without examining the inside circuitry. For this purpose, the power amplifier in figure 3-1 can be considered as black box. By just examining the input and the output of the power amplifier, the effect of the PA can be estimated, no other analyses is needed in black box approach. This allows users to model a PA for non-linearities and memory effects by just using simulations, after the extraction of input and output. Which can be seen as an eased analyze of the PA. Only with single test input and output, one can model the PA and after that can examine the response of the PA for any input signal which has same bandwidth.

Due to the effects of non-linearity and memory of PA, methods that are capable of representing these effects are needed. In this paper Volterra series based modelling methods are implemented and tested. The details about the implemented five different Volterra based methods can be found in chapter 2.

Modelling results are compared according to two criterias, normalized mean square error and coefficient number (complexity). Depending on the application one can decide according to these results. Increasing the coefficient numbers not necessarily means better performance, however most of the cases this assumption is valid. It is designer's decision to choose the appropriate performance – complexity match.

3.2 Measurement Setup and Test Signals

All measurements for the modelling purposes have been done with the carrier frequency of 2140 MHz (band 1-mid uplink [2]). To check the performances of the modelling methods, 256 Quadrature Amplitude Modulation (QAM) Long Term Evolution (LTE) signal have been used as test inputs, which can be seen in figure 1-3. It has a bandwidth of 25 MHz, it consists of 2 cascaded 5 MHz LTE signal with a spacing of 15 MHz.

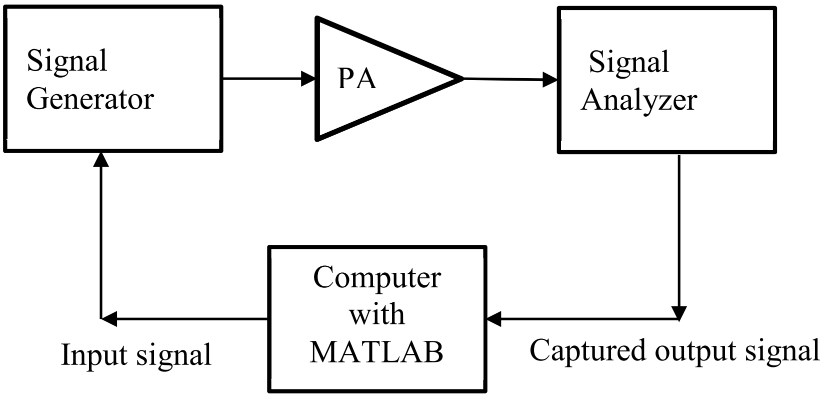

Figure 3-2: Schematic of the measurement setup

The measurement setup can be seen in figure 3-2. To be able to capture the data correctly and alleviate aliasing, same sampling rates are used in the spectrum analyzer and the signal generator. The spectrum analyzer has a maximum sampling frequency of 106.25 MHz with 85 MHz of capturing bandwidth, so the test signals are also created with this sampling rate and the signal generator is set for the same sampling rate. The input signal is sent from signal generator by using MATLAB and output of the PA is extracted from the spectrum analyzer in MATLAB. By using the output signals, each power amplifier is modelled.

3.3 Modelling Method

After extracting the output signals for corresponding input signals, the output signals have to be aligned before modelling the power amplifier. Alignment is done by calculating the cross-correlation between the output (received) signal and input signal. Due to input signal is sent continuously figure 3-4, by using cross correlation results, the alignment can be done. After the alignment, input and received output signal is used for modelling the PA. Signals in time domain before and after the alignment can be seen in figure 3-5 and figure 3-6.

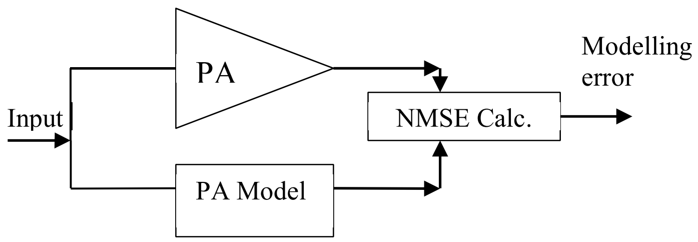

Figure 3-3: Schematic of the comparison method after modelling the PA

After the alignment, input and output signals are used to estimate the coefficients of the 5 modelling methods that is described in chapter 2. Least square estimation method is used for this purpose and it will be explained in detail in the next part.

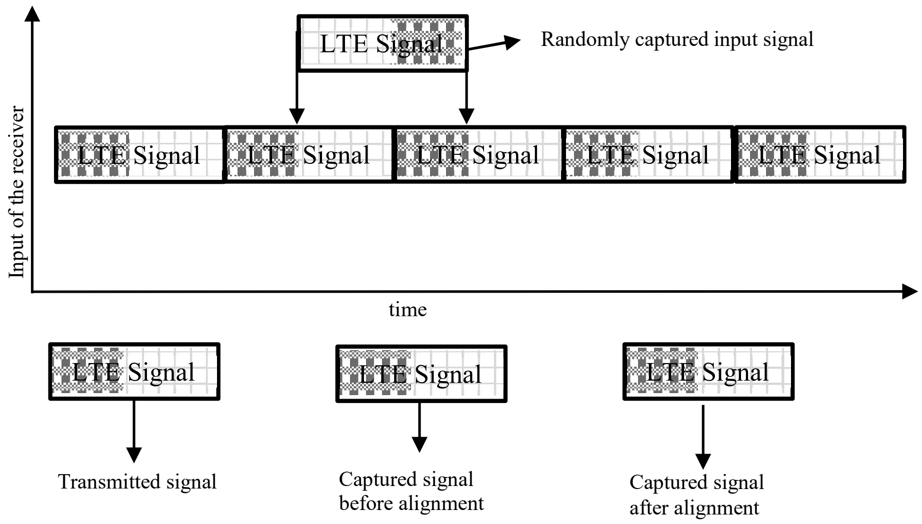

Figure 3-4: Diagram of received and captured signal before and after alignment

After estimating the coefficients for each method, in MATLAB environment these coefficients are used as power amplifier models i.e. input signals are fed to these PA models and the output of these models are compared with the received output from the signal analyzer which can be seen in figure 3-3.

In MATLAB environment, modelling parameters are also swept to get the best performance results. In the results section these comparison results for different PAs will be shown.
The comparison is done by Normalized Mean Square Error (NMSE) which can be described as:

$$NMSE_{db} = 10log_{10}\left[\frac{\sum_{n=1}^{N}|y_{meas(n)} - y_{model}(n)|^2}{\sum_{n=1}^{N}|y_{meas}(n)|^2}\right]$$

3-1

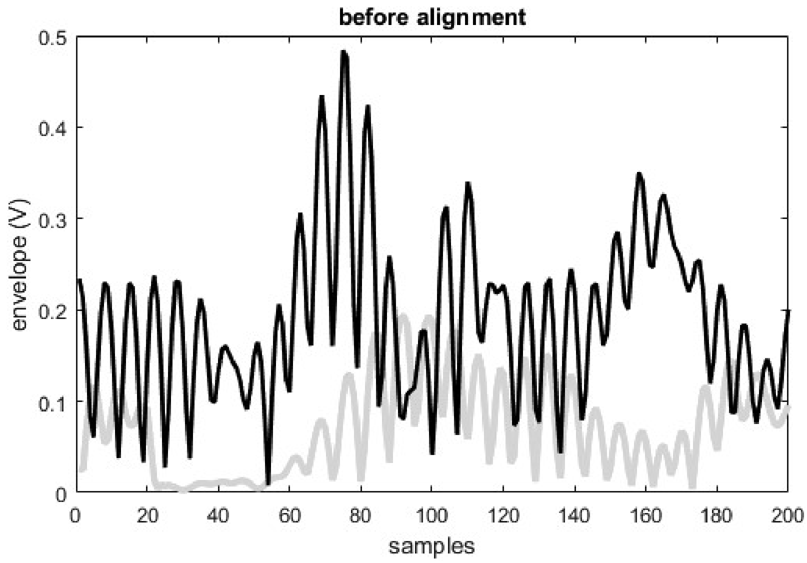

Figure 3-5: input and captured output before alignment

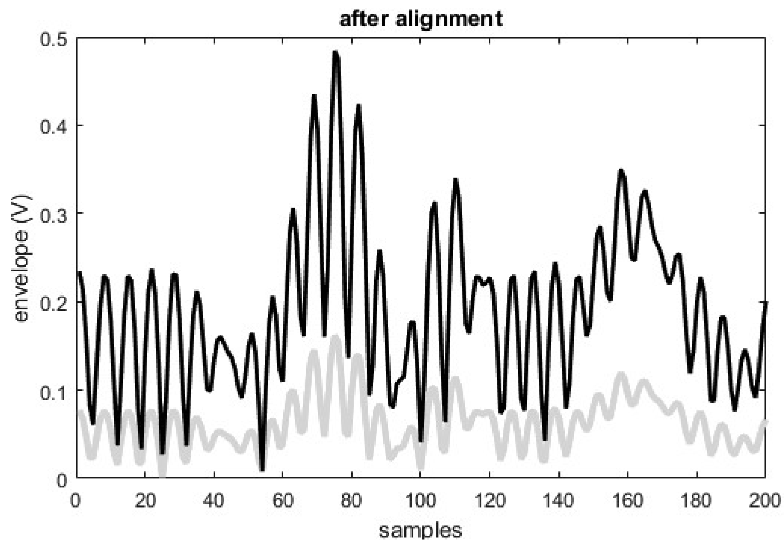

Figure 3-6:input and captured output after alignment

3.3.1 Coefficient Estimation by using Least Square

For the Volterra series based modelling methods that are used in this project, the coefficients are the key values that define each PA. Each PA has different coefficients and this difference creates the uniqueness of PAs in terms of models. For all the methods that has been explained in chapter 2, Least squares algorithm is used for model identification (coefficient estimation). However as mentioned before, for MP, GMP, SV methods single loop estimation algorithm and for ACR-GMP and ACR-SV multiple loop estimation algorithms are used.

Coefficients are estimated by minimizing a least square criterion *J*, calculated on a set of input and output signals which can be described as [20]:

$$J = \sum_{n=0}^{n=S-1} |e(n)|^2 = \sum_{n=0}^{n=S-1} |x(n) - z(n)|^2$$

3-2

Here *x* is the output and *z* is the output of the PA model. By using *x* as *z*, *J* can be minimized which leads to estimated coefficients with minimum error. Before describing the method in detail, first the definition of Volterra terms should be given.
If 2.7, 2.8, 2.9 is examined one can see that for each *k, m, l* value there is a corresponding coefficient. If these coefficients are written in a row vector, then the rest of the multiplication terms which is called Volterra terms can be written in a matrix. By multiplying this row vector with the matrice, the result of the equation can be found. An example for MP for power order K and memory order M for an input signal *x* with the length of N is as follows:

$$y_{MP\,(1\times\mathrm{N})} = [\;a_{10} \quad a_{11} \quad \cdots \quad a_{1M} \quad a_{20} \quad \cdots \quad a_{KM}\,]_{\;1\times\mathrm{KM}} \quad \times$$
$$\begin{bmatrix} x(0) & x(1) & & x(N) \\ x(-1) & x(0) & & x(N-1) \\ \vdots & \vdots & & \vdots \\ x(-M) & x(1-M) & \cdots & x(N-M) \\ x(0)|x(0)|^1 & x(1)|x(1)|^1 & & x(N)|x(N)|^1 \\ \vdots & \vdots & & \vdots \\ x(-M)|x(-M)|^{K-1} & x(1-M)|x(1-M)|^{K-1} & & x(N-M)|x(N-M)|^{K-1} \end{bmatrix}_{\mathrm{KM}\times\mathrm{N}}$$

3-3

Equation 3-3 is an example of MP which is defined in equation 2.7. Here each column represents the Volterra terms of each sample. Also for GMP and SV method same logic can be applied. In a system if input signal x and output signal y is known, only unknown in the equation 3-3 is the coefficient vector. If the equation 3-3 is written in multiplication format:

$$y = A \times X$$

3-4

Where y is the captured output signal with the size of $1 \times N$, A is the row vector of coefficients with the size of $1 \times KM$, X is the Volterra terms of the input signal x with the size of $KM \times N$. Vector A (coefficients) can be estimated by using least squares estimation as:

$$A = (X^H X)^{-1} X^H y = X^+ y$$

3-5

Where $(.)^{\mathrm{H}}$ denotes the Hermitian transpose and $(.)^{+}$ Moore-Penrose pseudo-inverse. By using equation 3-5 all coefficients can be estimated to model a PA from the input and captured output signal for MP, GMP, SV methods with a single loop. The only difference for GMP and SV is that their Volterra terms will be different, which can be easily found by the definition equations of each method.

As it can be seen from their definition equations 2-10, 2-14 for ACR-GMP and for ACR-SV there is not a direct linear relationship between the coefficients and the Volterra terms. This makes the single loop estimation method invalid. These 2 methods use the same logic to estimate the coefficients. For ACR-GMP estimation which is shown in [18] is as follows: First, 2 sets of coefficients which are defined at 2-11 are written in vector format:

$$\mathbf{R} = [r_1^{(a)}, \dots, r_{K_a}^{(a)}, r_1^{(b)}, \dots, r_{K_b}^{(b)}, r_1^{(c)}, \dots, r_{K_c}^{(c)}]^{\mathrm{T}}$$

3-6

$$\mathbf{T} = \begin{matrix} [a_0, \dots, a_{M_a,} b_{01}, \dots, b_{0L_b}, \dots, b_{M_b,1}, \dots, \\ b_{M_b,L_b}, c_{01}, \dots, c_{0L_c}, \dots, c_{M_c,1}, \dots, c_{M_c,L_c}, d_0, \dots, d_{M_d-1}] \end{matrix}^{\mathrm{T}}$$

3-7

At the beginning of the estimation process R and T vectors are initialized as:

$$\mathbf{R} = [1,0, \dots ,0]^{\mathrm{T}}$$

3-8

$$\mathbf{T} = [1,0 \ldots,0,1,0 \ldots,0,1,0 \ldots,0,0 \ldots,0]^{\mathbf{T}}$$

3-9

In vector T; a_0, b_{01}, c_{01} values are initialized as 1 and the rest of the values initialized as 0. As the iteration goes on first R values are estimated in the first loop and after that T values are estimated in the second loop.

The estimation algorithm for both of the modelling methods ACRGMP and ACRSV are same, first the output of the NME sub-block is deembedded from the captured output waveform and the static non-linearity which is **R** is estimated based on the least squares algorithm. Then, when **R** is known, it is applied to the input waveform and the memory effect coefficients which is **T** are estimated based on the least squares algorithm. The details about the estimation algorithm can be found in [18].

The key point for both behavioral modelling and DPD purposes is the estimation of the coefficients, as only these coefficients depends on the PA.

4 Digital Predistortion

4.1 Digital Predistortion Basics

As mentioned in the introduction, PAs are one of the most important components in wireless communication systems. However, when the PA is driven in high efficiency modes (in compression regions), it becomes non-linear which violates the 3GPP [2] spectrum regulations. In the compression region the peak points of the output of the PA becomes clipped, which ruins the output frequency spectrum figure1-6, figure 1-7. By estimating the inverse of the PA and placing it before the PA, the clipping effect of the PA is tried to be compensated which is demonstrated in figure 4-1 i.e. the power amplifier tried to be linearized. However, in this linearization process memory effects of the PA also should be considered. With the increase in the bandwidth in communication systems such as wideband CDMA (WCDMA), wideband OFDM (W-OFDM) memory effects of the PA become more important and cannot be ignored in the linearization process. Linearization without considering the memory effects in high bandwidth communication, results in poor performances [15] [23]. The models that are explained in chapter 2, all considers the memory effects. These 5 models are tested with 4 different power amplifiers and with two 5 MHz cascaded LTE signals with 15 MHz spacing. Results are examined in adjacent channel power ratio (ACPR) and coefficient numbers (complexity). However, decreasing the ACPR level was the real target of this project, so during the research and testing process coefficient number was not considered. The number of coefficients are provided for a reader who is also interested in the complexity of each method.

If a power amplifier was ideal, the difference between the input and the output of the PA would be only the gain which is just multiplication with a scalar number. More over if the PA was ideal, the gain of the power amplifier would be same for all input levels, however due to gain compression which also causes non-linearities; the gain is not constant for all input levels. To compensate the non-linearity and memory effect of the PA, digital predistortion algorithm is needed. The block diagram of the solution that is examined in this paper is shown in figure 4-1 [24]. The purpose of the DPD, as mentioned before; to compensate the non-linear effects of the power amplifier to the signal without distorting the gain of the amplifier. For this purpose, the inverse transfer function of the PA is found and placed before the PA in the communication chain.

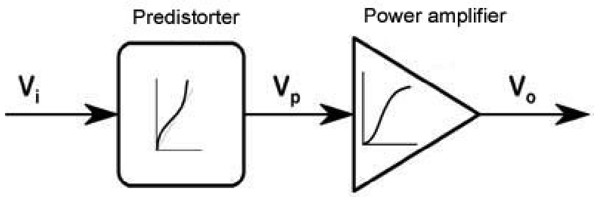

Figure 4-1: Inverse transfer function of a power amplifier placed before the power amplifier

4.2 Digital Predistortion Algorithm

4.2.1 Digital Predistortion

To be able to linearize the PA, non-linearities and the memory effect of the PA should be estimated from the captured data. One of the techniques for this operation is placing an inverter function after the PA at the output. This post-converter can be used as a pre-inverting model for the DPD [25]. This technique is called the *indirect learning architecture,* a block diagram of this technique which was proposed by [26] is shown in figure 4-2 which will be used also in this paper.

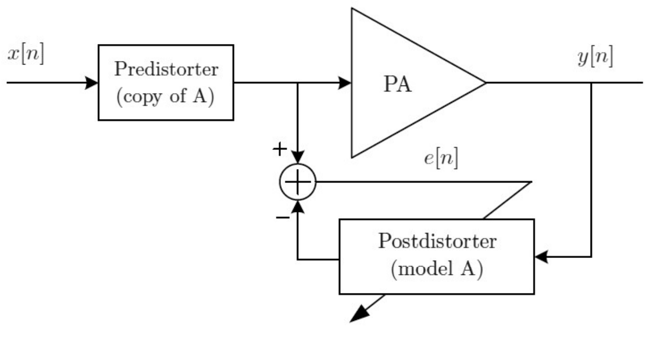

Figure 4-2: Basic schematic of digital predistortion of a power amplifier [26]

Estimation of the inverse of the PA is done by the postdistorter block. Exact same method that is used for behavioral modelling in chapter 3 is used to find the model A in figure 4-2, which is the inverse of the PA. Captured output of the PA considered as input and the input of PA is considered as output, so when the coefficients are estimated according to this combination the estimated coefficients are the coefficients of the inverse of the PA i.e. behavioral modelling of inverse of the PA. After that, the estimated PA inverse is placed before the PA. By iteratively estimating the inverse of the PA, final version of the postdistorter is found.

Postdistorter block can be used for digital predistortion for any signal with the same bandwidth for the specific PA. In other words, if a PAs postdistorter block is calculated with T number of iterations for a signal with bandwidth BW, any signal that has the bandwidth BW can be digitally predistorted by using the same postdistorter block without any iterations. Details about the postdistorter block can be found at the next chapter.

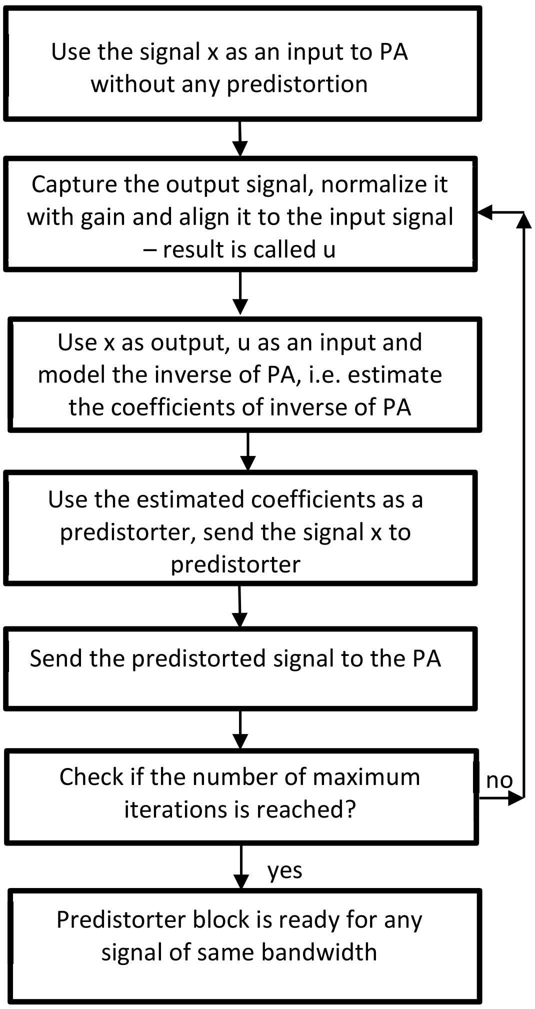

Figure 4-3: Flowchart of DPD algorithm

4.2.2 Postdistorter Block Estimation Algorithm

Any of the method that is explained in chapter 2 can be used to model the PA. Details about estimating the coefficients of each method can be found in chapter 3.

As mentioned before, if any of the Volterra series based modelling methods that are mentioned in this paper is used for modelling a non-linear system, only the coefficients of each method is unique to the non-linear system. In other words, in these methods only the coefficients are dependent to the system.

Flowchart of the DPD algorithm is shown in figure 4-3. Detailed explanation of the DPD algorithm is as follows.

Let the input signal be x[n] with the length of N. At the beginning of the estimation process, the input signal is directly send to the PA without any predistortion. The output signal y is captured with the length of N. After this step, captured signal should be gain normalized by the PAs gain so that the DPD algorithm can model the AM/AM and AM/PM characteristics of PA more accurately [27]. However, PAs does not have constant gains, their gains change with the input power – gain distortion figure 1-5, so choosing the gain value to normalize the signal is one of the key points. There are many different suggested methods in literature about choosing the normalizing gain of PA for DPD purposes. Some of the most commonly knowns are, normalizing to the maximum gain of the amplifier [28], normalizing to the gain at the maximum targeted output power [29] and normalizing the gain in a way that average power of the output of the predistorter does not change [30]. In this project a different method, is used for gain normalization which can be described as:

$$normalized\ output = captured\ output \times \left(\frac{mean(|input|)}{mean(|captured\ output|)}\right) \quad \mathbf{4.1}$$

In this method the root mean square (RMS) values of the input and the output signals are considered and the normalization of the output signal is done with these values. This method is used because of; RMS value is more reliable if the signals have high peaks (high peak to power ratio) and these peaks can mislead the normalization value.

After the normalization step, the input and the output signals should be aligned. Due to the cyclic signal transmission and also due to the longer capturing time then the signals duration, by using the correlation between the x and y, output signal y is aligned with the input signal x which can be found in chapter 3.3 in detail.

To find the inverse transfer function of the PA, aligned and normalized output signal y is treated as input signal and the input signal x is treated as output, using the behavioral methods that are explained in chapter 3, the inverse model is estimated. After this step the signal x is used as an input to the inverse model of PA (predistorter) then the output of the predistorter is sent to PA and the output is captured. By checking the number of iterations this process is done in loops. An adaptive algorithm can be implemented to check the ACPR level after each iteration, however in this project it was out of scope.

4.3 Measurement Setup

For the DPD measurements same setup in the behavioral modelling setup has been used, figure 3-2. By using MATLAB the signal generator and signal analyzer are controlled. The coefficients are estimated by using MATLAB. The signal that is used in DPD is the same signal which is used for behavioral modelling.

Before examining the results, it is important to briefly explain the drawbacks of the measurement system. During the tests, it is realized that the test setup has two main drawbacks. First of all, the signal generator and the signal analyzer has the maximum capturing bandwidth of 85 MHz, due to this specification when the behavioral modelling algorithms and DPD algorithms are tested bandwidths higher than 20 MHz couldn't be used. It is a fact that to model the PA for both purposes 3^{rd} and 5^{th} order intermodulation products have to be captured. To briefly explain, if one examines figure 1-8 the products in the fundamental zone depends on the bandwidth (due to LTE signals are not single tone) and the spacing between the different LTE signals. By increasing the bandwidth or the spacing between the signals, one can move the intermodulation products. However, to be able to model these effects one needs to capture these products. For the signal used in the tests for this project 3^{rd} and 5^{th} order intermodulation products are in the bandwidth of 85 MHz. The details about 3^{rd} order and 5^{th} order intermodulation products can be found in the introduction, and more details can be found in [10].

The second drawback of the setup that is observed is, the measurement devices -signal generator and signal analyzer- can create additional random noise which can distort the signal's low power points. This problem causes the modelling methods to model the effects of the PA poorly for low power levels. However, this random noise wasn't observed for each test, a pattern or the reason couldn't be found. More comments on these drawbacks and how to improve them can be found in chapter 7 -future work-.

5 Results

In this chapter both the forward behavioral modelling and the DPD results are shown. The last part of each subsection is the comments about the results.

5.1 Modelling Results

To check the performance of the methods for forward behavioral modelling, 2 different PAs have been used. Both PAs are driven in the compression region so that, the non-linearities can be observed and the performance of the methods to model non-linearities can be evaluated. For each PA and for each method the parameters are swept, the results of this sweeping is shown in corresponding graphs. A relation between parameters and behavioral modelling tried to be found. The table 5-1 shows the range of the swept parameters. The sweeping is stopped when the NMSE result started to converge to it's limit. Each method's NMSE value converges to its limit after a certain level of coefficients which can be seen on each PAs corresponding NMSE vs coefficient number graphs.

Model		MP	GMP	SV	ACRGMP	ACRSV
Parameter	Ka	1--6	1--6	2,4,6	2,4,6	2,4,6
	Kb		1--6	2,4,6	4,5,6	2,4,6
	Kc		1--6	2,4,6	4,5,6	2,4,6
	Kd			3		3--4
	Ke			3		3--4
	Ma	1--5	1--5	1--5	1,3,5	1,3,5
	Mb		1--3	2--3	2--4	2--4
	Mc		1--3	2--3	2--4	2--4
	Md			2--3	1--3	2--4
	Me			2--3		2--4
	Mf			2--3		1--3
	Lb		1--3	2--3	2--4	2--4
	Lc		1--3	2--3	2--4	2--4
	Ld			3		1--3
	Le			3		1--3

Table 5-1: Range of parameters that are swept for examining each method's behavioral modelling performance

5.1.1 Results of PA from Vendor 1

In this section results of the behavioral modelling for PA from vendor 1 will be shown. In figure 5-1, the NMSE vs coefficient number graphs for each modelling method is shown. As one can see from the graph each method reached its limit. The results can be seen on table 5-2. The best result is achieved with SV at -45.0 dB by using 204 coefficients. However, if these methods will be used in a project which resource usage is important, one can see that ACRSV method reaches -44.0 dB NMSE by using 84 coefficients, i.e. 1 dB less performance but saving 60 percent of the resources. There can be some applications which 1 dB NMSE increase is important, in that case one should use SV method for this PA. These results show that for PA number 1, none of the methods are best in terms of both NMSE and coefficient criteria, this creates a tradeoff, one should decide according to the projects requirements.

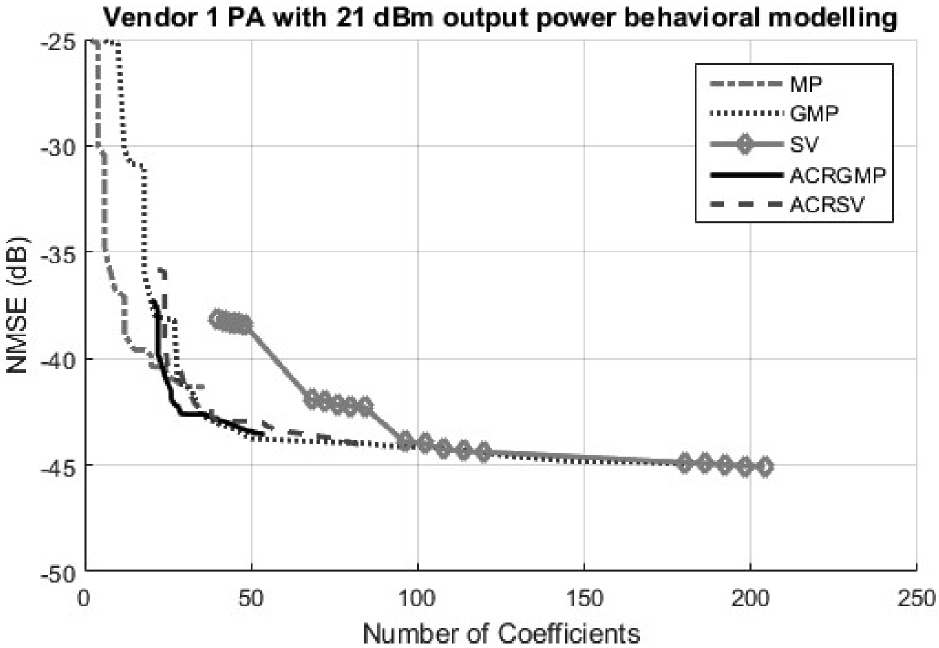

Figure 5-1:NMSE vs Coefficients number for 5 different methods, for behavioral modelling

Model		MP	GMP	SV	ACRGMP	ACRSV
Coefficient number		36	180	204	54	84
NMSE (dB)		-41,3	-44,9	-45,0	-43,5	-44,0
Dimension	**Ka**	6	6	6	6	6
	Kb		6	6	6	6
	Kc		6	6	6	6
	Kd			3		3
	Ke			3		3
	Ma	5	5	5	3	5
	Mb		3	3	4	4
	Mc		3	3	4	4
	Md			3	1	4
	Me			3		4
	Mf					3
	Lb		3	3	4	4
	Lc		3	3	4	4
	Ld			3		3
	Le			3		3

Table 5-2:Best NMSE results for each method and corresponding parameter values

5.1.2 Results of PA from Vendor 2

In this section results of the behavioral modelling for the PA from vendor 2 will be shown. In figure 5-1, each methods performance of behavioral modelling of the PA is shown. As it can be seen from the same figure, increasing the orders of the system, will not necessarily increase the performance of the modelling. From the graph it is understood that ACRGMP and ACRSV methods are superior methods in terms of modelling. Even though, ACRSV has slightly better performance than ACRGMP, the coefficient number to achieve this performance is 50 percent larger. The best values of each method for PA from vendor 2 with their corresponding parameters can be found at table 5-3. Moreover, the figure 5-2 shows that each method has a certain limit that can be achieved by increasing the orders of the method and when the method reaches its limit increasing the orders is useless.

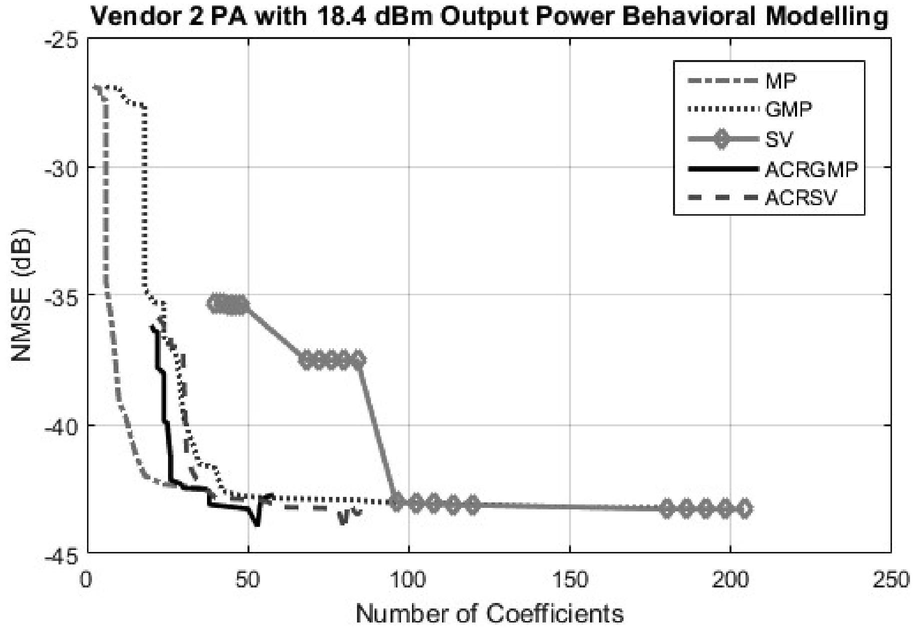

Figure 5-2: Number of Coefficients vs NMSE for 5 different behavioral modelling methods

Model		MP	GMP	SV	ACRGMP	ACRSV
Coefficient number		36	180	204	53	80
NMSE (dB)		-42,4	-43,2	-43,2	-43,9	-44,1
Parameters	**Ka**	6	6	6	6	6
	Kb		6	6	6	6
	Kc		6	6	6	6
	Kd			3		4
	Ke			3		4
	Ma	5	5	5	1	1
	Mb		3	3	4	4
	Mc		3	3	4	4
	Md			3	2	4
	Me			3		4
	Mf					1
	Lb		3	3	4	4

	Lc		3	3	4	4
	Ld			3		3
	Le			3		3

Table 5-3: Best behavioral modelling results and corresponding parameters for power amplifier from vendor 2

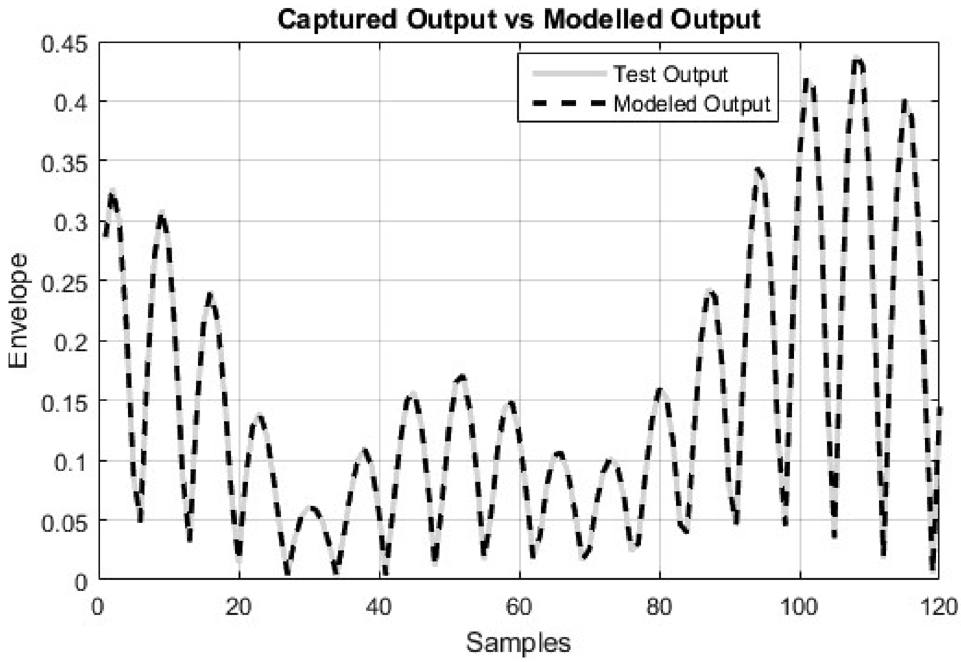

Figure 5-3: Measured output vs modelled output for ACRSV method

In figure 5-3, the captured output from the spectrum analyzer and the modelled output is shown. The method that is used here is ACRSV due to it has the best NMSE result compare to the other 4 methods. Also the results of behavioral modelling in frequency domain can be seen in figure 5-4.

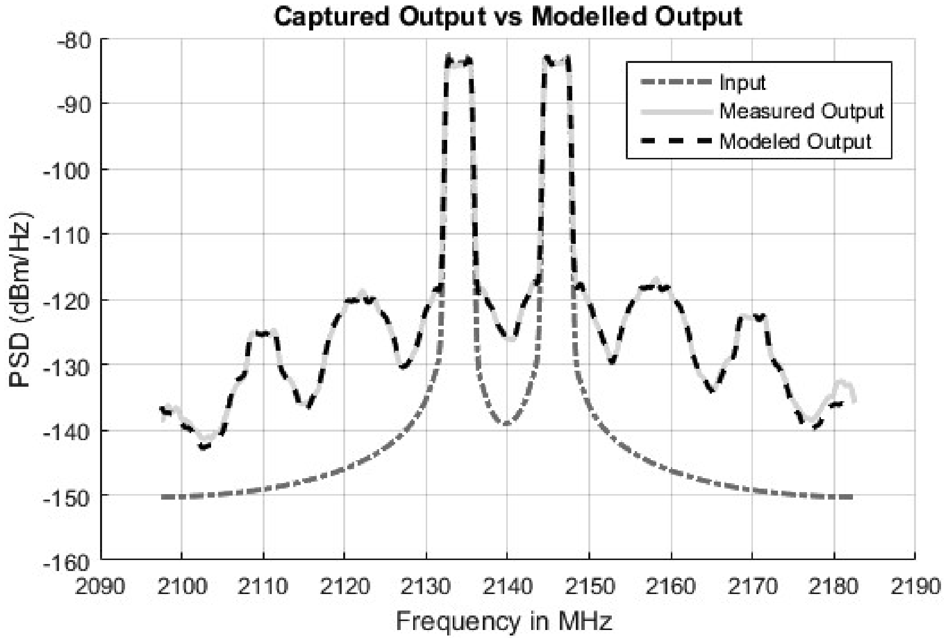

Figure 5-4: Input, captured output and the modelled output for ACRSV method

5.2 Digital Predistortion Results

The written 5 different predistortion algorithm was tested on 2 different PAs from 2 different vendors. Also, AM/AM and AM/PM curves for each PA test is shown. The purpose of this project is to compare these 5 methods for different PAs and if there is a superior method highlight it. However, the tests showed that for each PA a different method gives the best results.

Different output power levels are used for 2 different PAs due to their different characteristics however these different power levels have the same backoff from the 1 dB compression points. The power level is chosen according to the compression point, the purpose in these tests to run the PAs in compression region. The signal that is used for these tests has a non-constant envelope, so the total output power adjusted such that, the peaks of the signal are in the compression region. It is a known fact that if the PA is in 3-4 dB compression region linearization is not possible. So these power levels are adjusted by checking the results and highest possible power levels are used for each PA.
Moreover, the power levels in frequency spectrum plots have different levels due to the attenuators at the input of spectrum analyzer. The important values

in those plots are the ACLR levels which are ratios of the signal to adjacent channels which are not affected by the attenuators.

In all DPD tests that are done in this project the same parameters have been used. These parameters can be found at table 5-1, as mentioned before the resource usage -coefficient number- is not in the scope of this project. When deciding these parameters, for the common parameters between different methods the same values has been chosen and for the non-common parameters reasonable values from the reference papers have been used. One can find the definition of the parameters by checking their corresponding equations.

	MP	GMP	SV	ACRGMP	ACRSV
Ka	6	6	6	6	6
Kb	0	5	5	6	6
Kc	0	5	5	6	6
Kd	0	0	3	0	3
Ke	0	0	3	0	3
Ma	5	5	5	5	5
Mb	0	2	2	3	3
Mc	0	2	2	3	3
Md	0	0	2	1	2
Me	0	0	2	0	2
Mf	0	0	0	0	1
Lb	0	2	2	3	3
Lc	0	2	2	3	3
Ld	0	0	2	0	2
Le	0	0	2	0	2

Table 5-4: Parameters of DPD algorithms

5.2.1 Results of PA from Vendor 1

The DPD algorithm is tested for the first PA with an output power of 21 dBm.

If one examines figure 5-5, 2 methods are superior than the other methods. ACR-GMP and ACR-SV provides better suppression in ACLR1 levels. In table 5-5, ACLR levels are shown in dB. Without any DPD algorithm, the ACLR levels of the output signal is -36.7 dB and -34.23 dB which are higher than the specifications of 3GPP for LTE.

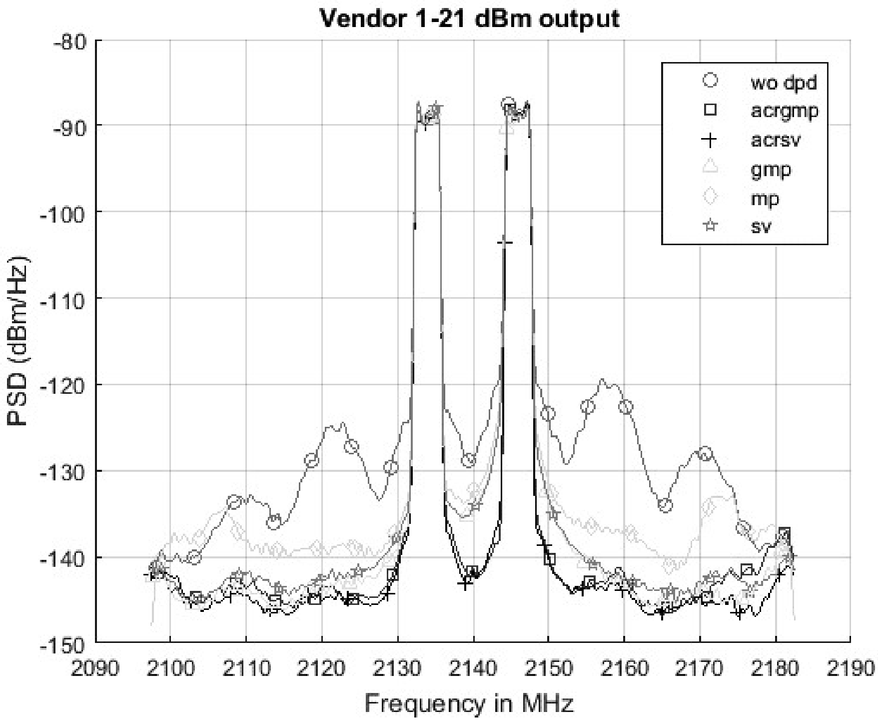

Figure 5-5: Frequency spectrum of output of the PA from vendor 1 after DPD, for 21 dBm output power

Method	Lower ACLR1	Higher ACLR1	Coefficient Number
w/o DPD	-36,7	-34,2	
MP	-46,1	-40,3	36
GMP	-48,9	-42,3	72
SV	-47,3	-42,2	108
ACRGMP	-49,5	-49,1	42
ACRSV	-50,4	-50,0	52

Table 5-5:ACLR results of figure 5-4

LTE standard limits the maximum ACLR1 -45 dB. MP, GMP and SV methods satisfies the LTE standards for lower ACLR1 however they fail for higher ACLR1. ACRGMP and ACRSV they satisfy both the higher and lower ACLR1. The reason for ACRGMP and ACRSV's superiority can be explained by the additional non-linear memory effect (NME) block. Because, if the equations of 5 different methods are examined, ACRGMP and GMP, ACRSV and SV are based on the same equations, the details can be found in their corresponding sections. The reason of ACRSV's superiority to ACRGMP is the additional cross terms which comes from complex conjugate multiplication elements which shown in equation 2-15. This result shows that for this specific PA, ACRSV method should be used for linearization purposes. To understand the effect of DPD algorithm to the signal sample points, the AM/AM and AM/PM curves can be examined. Figure 5-6 shows the AM/AM and AM/PM curves before and after the DPD algorithm. The high deviation at the low output power levels caused by the memory effects and the noise effect of the measurement setup that is mentioned as a drawback earlier. As you can see after the DPD algorithm the effects of memory is decreased and more over the distortion that can be seen at higher power levels is also compensated.

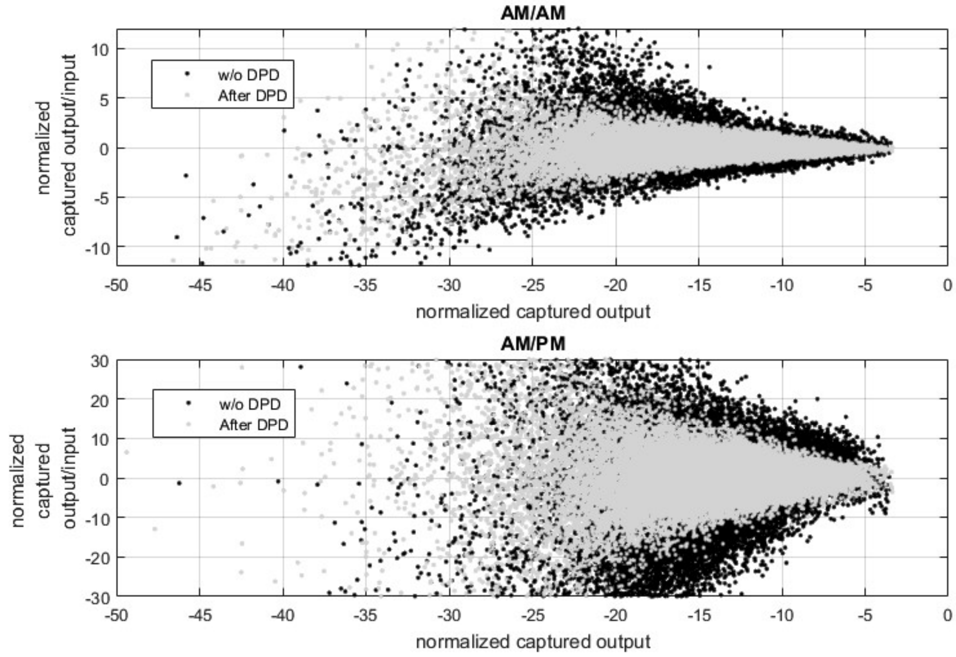

Figure 5-6:Comparison of AM/AM and AM/PM curves without DPD and after DPD

If the forward behavioral modelling results and the DPD results are compared for this PA, it is observed that superior behavioral model is not the superior DPD. For the PA from vendor1, best forward behavioral modelling achieved by SV and best DPD result is achieved by ACRSV. Although, the difference between the behavioral modelling results do not differ from each other much, still the best behavioral modelling method and best DPD method are not the same. However, coefficient number can be an important criterion if the system that is using the algorithm has limited memory. Then, it becomes a tradeoff between performance vs resource usage. One should recall that DPD is the same algorithm with forward behavioral modelling except the DPD models the inverse of the PA, by using exact same algorithms.

5.2.2 Results of PA from Vendor 2

The second PA's DPD test results are as followed. For this PA 18.4 dBm output power is used which has the same backoff from the 1 dB compression point as first PA. Without any changes in the setup or the algorithm the PA from vendor 1 is replaced with the PA from vendor 2, and the results are as follows:

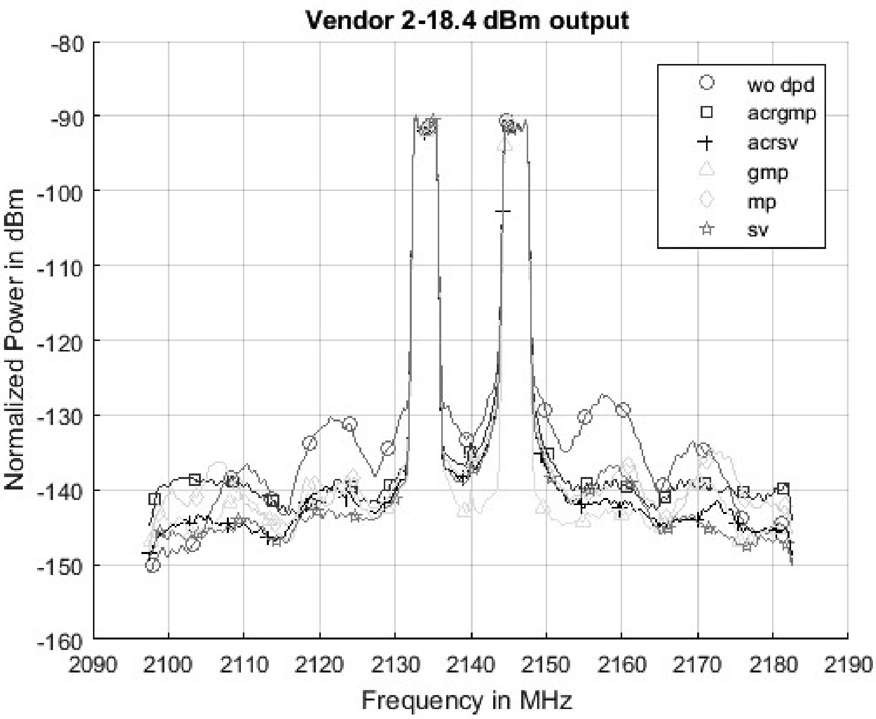

Figure 5-7: Frequency spectrum of output of the PA from vendor 1 after DPD, for 18.4 dBm output power

In figure 5-7, the frequency spectrum of the output is shown. For PA from vendor 2, the GMP model gives the best result in terms of ACLR1. The values of ACLR1 can be found at table 5-6.

The performance of GMP method shows us that for this PA, the additional cross terms with complex conjugate of the signal decreases the inverse modelling of PA, this is the reason why GMP outperform both SV and ACRSV. GMP's better performance than MP shows that cross leading and lagging terms in the method increases the inverse modelling performance. Lastly, when the performance of GMP and ACRGMP compared, the reason of GMP's better performance can be caused by additional NME block in ACRGMP, for this PA, NME block models the memory non-linearity poorly. The block that gave advantage for PA 1, can create a disadvantage for PA 2, due to their different characteristics.

Method	Lower ACLR1	Higher ACLR1	Coefficient Number
wo DPD	-39,4	-36,9	
MP	-46,2	-42,1	36
GMP	-48,5	-50,0	72
SV	-48,0	-43,1	108
ACRGMP	-46,1	-41,0	42
ACRSV	-47,4	-43,3	52

Table 5-6:ACLR1 results of Figure 5-3

In figure 5-4, the effect of DPD to AM/AM and AM/PM curves are shown. It is easily seen that, the DPD algorithm linearizes the peak points which are at the compression region of the PA for both amplitude and phase, which explains why the ACLR1 levels after DPD is lower.

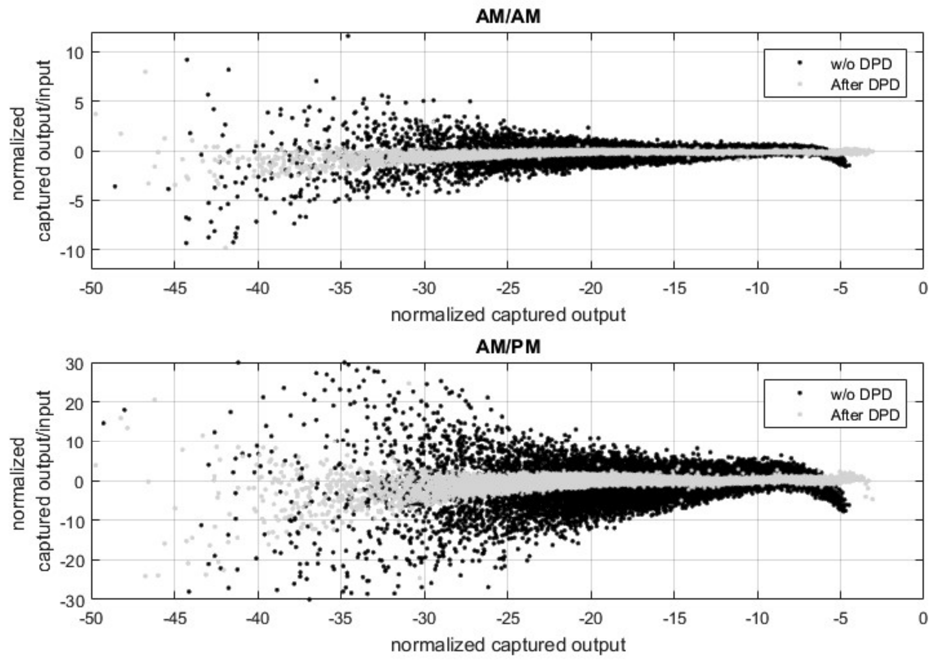

Figure 5-8: Comparison of AM/AM and AM/PM curves without DPD and after DPD

5.2.3 Comments on Results of 2 Different Power Amplifiers

If one compares the figure 5-6 and figure 5-8, one can see that in figure 5-6 there is a huge noise effect in the measurement setup. These 2 PAs have really similar noise characteristics when their datasheets are examined. This leads to the result of signal generator or signal analyzer randomly adds noise to measurement. It also shows that, the noise does not affect the ACLR compensation of the DPD algorithm however, it affects the AM/AM and AM/PM distortion. In other words, with low noise levels, memory effects of the PA compensated better.

Another important result of these tests is that; even though DPD algorithm uses the inverse behavioral modelling, for both tested PAs the best forward behavioral modelling method is not the best DPD method. If one examines the results of forward behavioral modelling methods and DPD results one can see that different methods gives the best results. This shows that by checking the behavioral modelling results one cannot decide to which DPD algorithm to use.

These results show that, for any PA one should first run tests and decide which method gives the best result and then create the lookup table for this method. This report concluded as there is not any superior method out of these 5 methods for the PAs that are tested, it is designer's responsibility to test and find the suitable method.
As mentioned before, a lookup table is only valid as long as PAPR level of the input signal and the bandwidth of the input signal is unchanged. After finding the best method for different bandwidths and PAPR levels, a lookup table for each corresponding signal can be created. These lookup tables should have the estimated coefficients of the decided method. These coefficients are estimated in loops as explained in chapter 4, however, the lookup table's coefficients should be the coefficients when the ACLR values converges to it's limits. When the lookup tables are prepared there is no need for any more iterations, each input signal will be fed into the inverse transfer function of the PA; which is done by using the coefficients from the lookup table and the corresponding method's equation. After that, these signals can be sent to the PA as an input.

6 Conclusion

To achieve the high efficiency levels, linearization techniques are required. Deciding which method to use for digital predistortion can be a hard process, due to the variety in DPD methods. For this project 5 different Volterra series based method, both for forward behavioral modelling and for DPD, is examined. 2 different power amplifiers from 2 different vendors are tested.

When the results are examined for forward behavioral modelling, one can see that each method reaches a convergence point. For the PA from vendor 1, even though SV method gives the best result, the resource usage is extremely high in compare to other methods.

For the PA from vendor 1, the best behavioral modelling result is achieved with SV -45.0 by using 204 coefficients. For the same PA if ACRSV method is examined, modelling NMSE is -44.0 by using 84 coefficients.1 dB worse performance but saving 60 percent of the resources. As mentioned before this creates a tradeoff between performance and resource usage.

For the PA from vendor 2, the best behavioral modelling result is achieved with ACRSV -44.1 dB by using 80 coefficients. However, ACRGMP methods reaches NMSE value of -43.9 with 53 coefficients. ACRGMP's performance is 0.2 dB less than ACRSV however with 33 percent less memory usage. For behavioral modelling of this PA, ACRGMP can be considered as best result due to the NMSE value and the resource usage.

When all the results of DPD algorithms are examined, for PA from vendor 1, ACRSV method gives the best result in terms of ACLR levels. It reaches up to -50.5 dBc on both higher and lower sides by using 52 coefficients. However, ACRGMP reaches 1 dB less ACLR by using 42 coefficients. For this PA it can be easily said that ACRGMP and ACRSV performs better than the other 3 methods both in terms of ACLR and resource usage. It shows that augmented methods, instead of linear coefficients works better for PA from vendor 1. Nevertheless, same conclusion cannot be made for the second PA that is tested.

When the results of PA from vendor 2 is examined, except GMP method all other methods fail to fulfill the requirements of 3GPP for 4G. However, GMP method gives a result of -48 dBc at the lower side and -50 dBc at the higher side by using 72 coefficients. Under these circumstances it is easy to say to be able to use this PA in the market, out of these 5 methods only GMP can be used for linearization purposes. This can be explained by, when SV and ACR-SV methods are used the complex conjugates of the input signals are considered in the modelling. However, if the PA does not have an effect which can be represented by complex conjugates this can decrease the performance of the modelling which will decrease the performance of the DPD. GMP method's better performance than MP can be explained by additional cross terms. Moreover, GMP's superior performance than ACR-GMP shows that separating non-linear memory block can decrease the performance of the system.

As mentioned at section 5.2.3, these results shows that for DPD, to be able to decide which method to use for a PA, tests should be done and then a decision should be made.

Moreover, if one compares the best behavioral modelling method and best DPD method for each PA, one can see that there is no direct connection.

Throughout the project, it is shown that for behavioral modelling methods, there is not any absolute best method. For both of the power amplifiers, behavioral modelling methods created a tradeoff between performance and resource usage. For digital predistortion results, for PA number 1, ACRGMP and ACRSV outperform the other 3 methods. The decision to choose which one to use again depends on the resources. However, for PA number 2 only one method, GMP, satisfies the 3GPP requirements.

To conclude, to be able to decide which DPD method to use for a specific PA tests should be done. According to these tests one can decide which method is superior then the others. After deciding the method coefficients can be saved in a lookup table and used the PA as long as the PAPR characteristics and the bandwidth of the signal do not change. If the system uses different bandwidth signals, different lookup tables should be created for each bandwidth.

7 Future Work

During the project many points which could be further research are detected. First of all, as mentioned before the measurement devices that are used for this project showed some uncertainty by adding noise. The same measurements can be done with a better measurement system. This can improve the performance of DPD algorithms.

Moreover, due to the limited bandwidth in the measurement setup, performance of the DPD algorithms for higher bandwidth signals couldn’t be tested. With a higher bandwidth setup one can test the DPD algorithms for higher bandwidth signals, some adjustment for the parameters of each model can be needed, when the bandwidth increases the memory depth of the power amplifier increases.

Because of limited time, only 2 different PAs could be tested for this project. Testing different PAs can give better understanding for the performance of DPD algorithms.

Another improvement can be achieved by sweeping the parameters of DPD algorithms, for this project reasonable values for DPD parameters have been chosen according to the behavioral modelling results and according to the reference papers. However, by sweeping the parameters some improvement in ACLR levels can be achieved.

Lastly, as mentioned before there are some algorithms suggested in literature to estimate the parameters of the DPD algorithm. These can be also applied for the methods that are tested here.

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