Digital predistortion of semi-linear power amplifier

Examensarbete utfört i Datatransmission
av

Robert Karlsson

LITH-ISY-EX--04/3545--SE
Linköping 2004
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Handledare: Jan Arnsby, FOI
Examinator: Lasse Alfredsson, ISY
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In this thesis, a new way of using predisortion for linearization of power amplifiers is evaluated. In order to achieve an adequate power level for the jamming signal, power amplifiers are used in military jamming systems. Due to the nonlinear characteristic of the power amplifier, distortion will be present at the output. As a consequence, unwanted frequencies are subject to jamming. To decrease the distortion, linearization of the power amplifier is necessary.

In the system of interest, a portion of the distorted power amplifier output signal is fed back. Using this measurement, a predistortion signal is synthesized to allow suppression of the unwanted frequency components. The predistortion signal is updated a number of times in order to achieve a good outcome. Simulations are carried out in Matlab for testing of the algorithm.

The evaluation of the new linearization technique shows promising results and that good suppression of distortion components is achieved. Furthermore, new predistortion features are possible to implement, such as predistorsion in selected frequency bands. However, real hardware testing needs to be carried out to confirm the results.
Abstract

In this thesis, a new way of using predisortion for linearization of power amplifiers is evaluated. In order to achieve an adequate power level for the jamming signal, power amplifiers are used in military jamming systems. Due to the nonlinear characteristic of the power amplifier, distortion will be present at the output. As a consequence, unwanted frequencies are subject to jamming. To decrease the distortion, linearization of the power amplifier is necessary.

In the system of interest, a portion of the distorted power amplifier output signal is fed back. Using this measurement, a predistortion signal is synthesized to allow suppression of the unwanted frequency components. The predistortion signal is updated a number of times in order to achieve a good outcome. Simulations are carried out in Matlab for testing of the algorithm.

The evaluation of the new linearization technique shows promising results and that good suppression of distortion components is achieved. Furthermore, new predistortion features are possible to implement, such as predistortion in selected frequency bands. However, real hardware testing needs to be carried out to confirm the results.

Sammanfattning


Utvärderingen av tekniken visar att ett gott resultat kan uppnås och att undertryckningen av distorsionskomponenterna är bra. Dessutom möjliggörs nya sätt att använda predistorsion, bland annat kan minskad distorsion i utvalda frekvensband erhållas. För att konfirmera resultaten måste dock hårdvaruimplementering och testning ske.
Preface

This work was carried out at the Swedish Defence Research Agency in Linköping. It has been very rewarding to work in close connection to technical research and to follow real life engineering.

I would like to thank my supervisor at FOI, Jan Arnsby, for giving me the opportunity to perform this thesis at FOI. I also would like to express my gratitude towards Per-Åke Andersson and Rolf Jonsson for valuable help. My opponent Fredrik Hasfjord should also consider himself thanked for proofreading and for executing an excellent opposition. Lastly, I would like to thank my Anna for proofreading and for, above all, believing in me.

Linköping, 11 November 2004

Robert Karlsson
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## Abbreviations and acronyms

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<thead>
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<th>Acronym</th>
<th>Written out</th>
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</thead>
<tbody>
<tr>
<td>FOI</td>
<td>Totalförsvarets Forskningsinsitut (Swedish Defence Research Agency)</td>
</tr>
<tr>
<td>PA</td>
<td>Power Amplifier</td>
</tr>
<tr>
<td>A/D</td>
<td>Analogue to Digital (conversion)</td>
</tr>
<tr>
<td>D/A</td>
<td>Digital to Analogue (conversion)</td>
</tr>
<tr>
<td>SP</td>
<td>Signal Processing</td>
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<tr>
<td>FFT</td>
<td>Fast Fourier Transform</td>
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<tr>
<td>DC</td>
<td>Direct Current</td>
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<td>DSP</td>
<td>Digital Signal Processing</td>
</tr>
<tr>
<td>IMD</td>
<td>Intermodulation Distortion</td>
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<tr>
<td>IMP</td>
<td>Intermodulation Products</td>
</tr>
<tr>
<td>AM/AM</td>
<td>Amplitude Modulation/Amplitude Modulation</td>
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<tr>
<td>AM/PM</td>
<td>Amplitude Modulation/Phase Modulation</td>
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<tr>
<td>RF</td>
<td>Radio Frequency</td>
</tr>
<tr>
<td>PD</td>
<td>Predistortion</td>
</tr>
<tr>
<td>PAE</td>
<td>Power Added Efficiency</td>
</tr>
<tr>
<td>ADC</td>
<td>Analogue to Digital Converter</td>
</tr>
<tr>
<td>DAC</td>
<td>Digital to Analogue Converter</td>
</tr>
<tr>
<td>rms</td>
<td>root mean square</td>
</tr>
</tbody>
</table>
1 Introduction

In this chapter an introduction is given to the thesis project. A background is presented and the problem of interest is identified. The objectives of the report are formulated and the outline is described.

1.1 Background

This Master thesis project has been carried out at the Swedish Defence Research Agency, FOI, in Linköping. FOI conducts research and technology development mainly towards the National Total Defence. The Department of Electronic Warfare Systems, where this work has been carried out, focuses on signal intelligence and jamming of radar and radio.

A current project at the department is STEJL, which combines the jamming and direction finder objectives of radio communication [1]. The system is able to locate and jam both fix frequency and spread spectrum signals at the entire communication bandwidth. This is achieved by using Field Programmable Gate Arrays as the core of the system, hence allowing a flexible and computationally fast implementation.

1.2 Problem identification

In all systems handling radio communication there is always signal amplification involved. Before feeding the communication signal to the antenna and transmitting it through the ether, the signal needs to be amplified. For this purpose power amplifiers are used, giving the signal the boost needed for communication.

However, when amplifying the signal, the power amplifier (PA) will introduce unwanted frequency components to the amplified signal. Instead of achieving a purely linear response, the PA will act as a nonlinear system. This is a well known problem in mobile systems, where spectral efficiency is essential and disturbance of co-channels means degraded system performance. Therefore a lot of effort is put into solving this problem worldwide. Improved electrical circuits and signal processing are two ways of improving linearity.

Another possibility to improve linearity is to reduce the input signal power. There is a trade off between linearity and energy efficiency in PAs; reduced input power results in increased linearity but lower efficiency. However, the need for high efficiency, especially in military applications, makes this alternative unattractive.
In the problem of interest, a jammer system is the transmitting (and receiving) part. The purpose of the jammer system is, simply put, to make sure that communication is not possible on the frequencies used by the enemy. Since the PA is nonlinear, energy will be transmitted at other frequencies as well. This could be very critical in a military communication environment where friendly channels may be jammed or specific emergency frequencies corrupted.

![Figure 1.1 Basic system description; signal processing unit, power amplifier, A/D and D/A conversion units and antenna. PA signal is fed back to SP. Input to SP selects which frequencies to jam.](image)

A basic description of the jammer system is presented in Figure 1.1. The SP (signal processing) unit contains different system components, such as signal generator, frequency analyzer and pre-amplifier. The input to the system is the frequencies selected for jamming ($\sum f_q$). The idea is to take a portion of the amplified signal and feed it back to the SP. The system contains FFT (Fast Fourier Transform) calculation circuits, allowing processing of signal information in the frequency domain. That is, the signal components are calculated with amplitude, frequency and phase. By using this information and simply adding the phase shifted unwanted signal component to the original signal, suppression of the nonlinear signals from the PA should be obtained.

The addition of more signals also means that new frequency components are added to the output due to the nonlinear nature of the PA. Therefore a loop is required that enables updating of the signal fed to the PA. By iterating a number of times a stable signal constellation should be derived and improved spectral properties achieved.

### 1.3 Purpose

The purpose of this work is to test whether the above mentioned idea is valid or not. If valid, an estimation of the improvements in terms of suppression of unwanted spectral components is desirable.

The algorithm will be developed, implemented and evaluated.

Before testing the algorithm, a model of the power amplifier needs to be synthesized. It should capture the essential features of the amplifier to make sure that the testing of the algorithm is not too far from reality.

The model and algorithm are implemented in MATLAB®.
1.4 Theory studies
To be able to understand and carry out testing, the theory for power amplifiers and predistortion was studied. All theory was found in books, papers and at internet sites. Some literature was provided at the start-up and some was found during the progress of the work. The predistortion algorithm was orally formulated, providing the basis for the implementation. This was also discussed with the supervisor at FOI during the implementation process.

1.5 Outline
Chapter 2 introduces the nonlinear theory of power amplifiers and describes different sources of distortion.

In Chapter 3, a number of modelling techniques are presented. The emphasis is on polynomial modelling and a description of one such particular technique is given.

Linearization is the subject in Chapter 4, briefly describing various approaches and introducing the predistortion technique of interest.

Chapter 5 shows the setup used for measurements and discusses these. Useful units and conversions are presented.

The results from the modelling and the simulations are found in Chapter 6.

Finally, conclusions and suggestions for further work are given in Chapter 7.
Digital predistortion of semi-linear power amplifier
2 Power amplifier nonlinearity

In this chapter the theory of distortion and nonlinearities in power amplifiers is presented. Amplitude and phase distortion as well as other nonlinear phenomena are characterized and discussed.

2.1 Amplitude distortion

Power amplifiers (PAs) are essential parts of radio communication systems. They boost the communication signal to adequate power levels before feeding it to the antenna and transmitting it through the ether.

Ideally, one would like to have linear PAs. The transfer characteristic for a perfect linear system is seen in Figure 2.1.

The input signal is amplified with a scalar quantity, producing a perfect replica with no distortion present.
Unfortunately, in this case, reality is not ideal. The amplifier is first of all not able to produce arbitrarily large signals and will be saturated. Furthermore, the transfer characteristic is not linear up to the saturation point. The amplification decreases as the input power increases. A transfer characteristic is more likely to look something like the dashed line in Figure 2.2.

![Figure 2.2 Comparison between an ideal (solid) and a nonlinear (dashed) transfer characteristic.](image)

A linear curve (solid) is plotted for comparison. The saturating and nonlinear behaviour is easily seen as the amplification decreases at higher input levels. There are many ways to express the nonlinear relationship mathematically. Thanks to the straightforward form and, as we shall see later, the connection to the frequency domain output, polynomials are often used in technical literature. If polynomials are selected for this purpose, a power series will describe the relationship, i.e.

\[ v_{\text{out}}(t) = a_1 v_{\text{in}}(t) + a_2 v_{\text{in}}(t)^2 + a_3 v_{\text{in}}(t)^3 + \ldots \]

The transfer characteristic now includes higher order terms, not only the linear term. In the above plot (Figure 2.2), a third order polynomial represents the nonlinear transfer function. The second-order coefficient is positive and the third-order coefficient negative, which result in a compressive characteristic of the curve. The more the input signal grows, the larger the influence of the higher-order powers.

When viewing the nonlinear response of the amplifier in the frequency domain, a number of extra frequencies will be noticed. To exemplify, let the input to the nonlinear system be a cosine with an arbitrary (angular) frequency, i.e.

\[ v_{\text{in}}(t) = \cos(\omega t). \]
The output will be (if we assume a third order nonlinearity)

\[ v_{out}(t) = a_1 \cos(wt) + a_2(\cos(wt))^2 + a_3(\cos(wt))^3 \]

By applying simple trigonometric relationships, the output looks like

\[ v_{out}(t) = a_1 \cos(wt) + \frac{a_2}{2} \left( \frac{1}{2} \cos(2wt) \right) + a_3 \left( \frac{3}{4} \cos(wt) + \frac{3}{4} \cos(3wt) \right) = \frac{a_2}{2} + (a_1 + \frac{3}{4} a_3) \cos(wt) + \frac{a_2}{2} \cos(2wt) + \frac{3}{4} a_3 \cos(3wt) \]

The output signal now includes a DC-term, the original cosine-signal, a cosine with double frequency and a cosine with three times the original frequency. An example of such a response is plotted in Figure 2.3. The original signal has a frequency of 20 Hz.

The frequencies which are multiples of the original frequency are referred to as harmonics. Second harmonic is twice the original frequency, third harmonic three times etc.

From the result, we conclude that when feeding an amplifier with a signal of some frequency, the output signal will also include unwanted frequency components. This is referred to as AM/AM distortion, since the output amplitude will be distorted in relation to the input amplitude.
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Two-tone analysis

An established way of characterizing amplifier nonlinearity is the two-tone test. The input signal to the PA consists of two carriers with separate frequencies. This is equal to an amplitude modulated signal as seen in the following expression:\(^1\)

\[
v_{in}(t) = v \cos(\omega_1 t) + v \cos(\omega_2 t) = 2v \cos\left(\frac{\omega_1 - \omega_2}{2} t\right) \cos\left(\frac{\omega_1 + \omega_2}{2} t\right)
\]

The spacing of the frequencies is generally much smaller than their individual frequencies. By varying the amplitude \(v\), the device under test (in this thesis a PA) may be examined at different power levels. If we assume that the transfer characteristic has the power series form presented above with a truncation after the fifth term, i.e.

\[
v_{out}(t) = a_1 v_{in}(t) + a_2 v_{in}(t)^2 + a_3 v_{in}(t)^3 + a_4 v_{in}(t)^4 + a_5 v_{in}(t)^5
\]

the response from a two-tone signal will be

\[
v_{out}(t) = a_1 v (\cos(\omega_1 t) + \cos(\omega_2 t)) + a_2 v^2 (\cos(\omega_1 t) + \cos(\omega_2 t))^2 + a_3 v^3 (\cos(\omega_1 t) + \cos(\omega_2 t))^3 + a_4 v^4 (\cos(\omega_1 t) + \cos(\omega_2 t))^4 + a_5 v^5 (\cos(\omega_1 t) + \cos(\omega_2 t))^5
\]

Expansion of this expression yields

\[
v_{out}(t) = a_1 v \cos(\omega_1 t) + a_1 v \cos(\omega_2 t) + a_2 v^2 + \frac{1}{2} a_2 v^2 \cos(2\omega_1 t) + \frac{1}{2} a_2 v^2 \cos(2\omega_2 t) + a_2 v^2 \cos(\omega_1 + \omega_2 t) + a_3 v^3 \cos(\omega_1 - \omega_2 t) + \frac{9}{4} a_3 v^3 \cos(3\omega_1 t) + \frac{9}{4} a_3 v^3 \cos(3\omega_2 t) + \frac{3}{4} a_3 v^3 \cos(3\omega_1 + \omega_2 t) + \frac{3}{4} a_3 v^3 \cos(3\omega_1 - \omega_2 t) + \ldots
\]

A complete table of all signal frequencies and their coefficients can be found below, from [2, chapter 7].

\(^1\) Example from Chapter 7 in [2].
Table 2.1 Two-tone distortion products, up to fifth degree.

<table>
<thead>
<tr>
<th></th>
<th>$a_1v$</th>
<th>$a_2v^2$</th>
<th>$a_3v^3$</th>
<th>$a_4v^4$</th>
<th>$a_5v^5$</th>
</tr>
</thead>
<tbody>
<tr>
<td>1 (dc)</td>
<td>1</td>
<td>9/4</td>
<td>25/4</td>
<td></td>
<td></td>
</tr>
<tr>
<td>$\omega_1$</td>
<td>1</td>
<td>9/4</td>
<td>25/4</td>
<td></td>
<td></td>
</tr>
<tr>
<td>$\omega_2$</td>
<td>1/2</td>
<td>2</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>$\omega_1 + \omega_2$</td>
<td>1/2</td>
<td>2</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>$2\omega_1$</td>
<td>3/4</td>
<td>25/8</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>$2\omega_2$</td>
<td>1/4</td>
<td>25/16</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>$3\omega_1$</td>
<td>1/4</td>
<td>25/16</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>$3\omega_2$</td>
<td>1/4</td>
<td>25/16</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>$2\omega_1 + 2\omega_2$</td>
<td>3/4</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>$3\omega_1 + \omega_2$</td>
<td>1/2</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>$3\omega_2 + \omega_1$</td>
<td>1/2</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>$4\omega_1$</td>
<td>1/8</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>$4\omega_2$</td>
<td>1/8</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>$3\omega_1 + 2\omega_2$</td>
<td>5/8</td>
<td></td>
<td></td>
<td></td>
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<tr>
<td>$3\omega_2 + 2\omega_1$</td>
<td>5/8</td>
<td></td>
<td></td>
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</tr>
<tr>
<td>$4\omega_1 + \omega_2$</td>
<td>5/16</td>
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<tr>
<td>$4\omega_2 + \omega_1$</td>
<td>5/16</td>
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<td>$5\omega_1$</td>
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</tr>
<tr>
<td>$5\omega_2$</td>
<td>1/16</td>
<td></td>
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<td></td>
<td></td>
</tr>
</tbody>
</table>

The sum and difference frequencies, for example $\omega_1 \pm \omega_2$, are called *intermodulation (IM) products*. We see that the higher order powers in the expression produce signal frequencies that are of lower order. To exemplify, consider column three in the table above. The third-order term contributes not only to the third-order products (i.e. $2\omega_1 \pm \omega_2$, $2\omega_2 \pm \omega_1$, $3\omega_1$, $3\omega_2$), but also to the original frequencies $\omega_1$, $\omega_2$. The fifth-order term affects, besides the fifth order products, the original frequencies as well as the third-order products. The second order term, on the other hand, does not affect the original frequencies, nor does the fourth order term. Instead, the fourth order term contributes to the second order products. In other words, all odd-order terms affect all odd-order products of equal or lower degree, including the original frequencies since they are of order one. The same goes for the even-order terms; they affect all even-order products of equal or lower degree. Thus, if adding a seventh order term to the expression above, the fifth and third order products would be affected as would the original frequencies.

The extra frequencies in the output spectrum can be generally expressed with the relation\(^2\)

$$f_{dpi} = m \cdot f_1 \pm n \cdot f_2$$

where $f_{dpi}$ is the frequency component due to the nonlinear terms

- $f_1, f_2$ are the original frequencies
- $n + m = \text{order of distortion. } m, n \text{ are integer values including zero.}$

\(^2\) Taken from Chapter 2 in [3]. Note that the frequencies of lower orders not are included in the expression.
Example for 3\textsuperscript{rd} order nonlinearity is given below.

\[
\begin{align*}
    f_{dp1} &= 3f_1 \\
    f_{dp2} &= 3f_2 \\
    f_{dp3} &= 2f_1 + f_2 \\
    f_{dp4} &= 2f_2 + f_1 \\
    f_{dp5} &= 2f_1 - f_2 \\
    f_{dp6} &= 2f_2 - f_1
\end{align*}
\]

In the frequency domain it will look like in Figure 2.4.

![Figure 2.4 Output spectrum of third degree nonlinearity with a two-tone input signal.](image)

Generally speaking, the most serious IM-products (IMPs) are the ones close to the carrier signals. They are called in-band \textit{intermodulation distortions} (IMD) and are products of the odd order powers. Only odd order products will produce in-band distortion, the even order frequencies are usually too large. Figure 2.5 is a plot of the first four zones in the frequency domain, taken from \cite[Chapter 2]{3}. The fundamentals are the original frequencies. A measurement of the level of nonlinearity is to compare the original signal level with the level of the strongest IM (usually, but not necessarily, the third order).
It is evident that the odd-order intermodulation lies within the fundamental zone together with the original signals. Products of the even order powers and the second order harmonics are in the second harmonic zone. If the amplifier is used within a narrow frequency band, the products of higher order can be filtered out and do not constitute a big problem.

Filtering is not a solution to suppression of the close-to-carrier IMs. It should be noted that in Figure 2.4 only the third order IMs are present, while in most cases there are higher degree components too, especially influential at high signal levels, making the problem worse. Furthermore, only two carriers form the input, whereas in for example mobile systems a number of frequencies are amplified.

2.2 Phase distortion

The conversion of input power to output phase is called AM/PM response. The amplitude of the input signal affects the output signal phase. Increasing amplitude levels will introduce an increasing phase distortion on the output signal. The AM/PM curve may look something like Figure 2.6. The figure is based on measurements later presented.

Any signal wave shape will encounter distortion in case the frequency components are not delayed the same time amount [3, Chapter 2]. The delay and phase shift are related as

\[ \tau = \frac{\phi}{2\pi f} \]

where \( \tau \) is the time delay in seconds, \( \phi \) is the phase in radians and \( f \) the frequency in Hertz.
If the phase is linearly proportional to the frequency, all frequency components will be delayed the same amount of time when processed in the amplifier.

![Phase response of a power amplifier.](image)

The AM/PM effect also contributes to the IMDs, making the amplitude of these larger. Generally, those contributions are lower in magnitude; the dominant effect is most often the AM/AM conversion [2, Chapter 7].

### 2.3 Other nonlinear phenomena

Memory effects in PAs are phenomena that are connected to frequency dependent behaviour. The transfer characteristic is not static for the PA, rather related to the bandwidth of the input signal (definition in [4, Chapter 2]). This means that the intermodulation products will be different at various bandwidths; not necessarily altering considerably. In [4] the memory effects are divided into two forms: thermal and electrical. No further analysis will be presented here. To put it simple: when the current signal is affected by previous signal levels, some kind of memory is present in the system.

The IMDs that are produced by the polynomial function have upper and lower frequencies of the same amplitude. This means that the magnitude of, for example, the two third order products being closest to the original signals are equally modelled. However, that is not the case in real PAs, where uneven components often are present. The difference is dependent on both frequency band and amplitude and is caused by a phase difference of the AM/AM and AM/PM effects [5, Chapter 3].

Another issue in PA behaviour is ageing and temperature. Ageing effects are naturally not erasable; improving circuit synthesis is a way of dealing with the effect. Temperature is a key issue in circuits, since PA properties can alter significantly with changing thermal environment. When electrical circuits are used they will inevitably produce heat.
which will affect the properties. Operating at higher ambient temperatures naturally means affected performance.

Spurious signals are signals which appear without a reasonable explanation [3, Chapter 2]. They might emerge and vanish randomly in time and frequency. The cause of them is badly manufactured and/or constructed hardware.

### 2.4 Efficiency issues

Efficiency in power amplifiers is a measurement of the relationship between the power that is fed to the circuit and what is actually received at the output (i.e. ratio of power fed and useful power achieved).

In Figure 2.7 below, the efficiency quantity PAE is used, which is calculated by using the following expression \(^3\),

\[
\text{PAE} = \frac{P_{OUT} - P_{IN}}{P_{dc}}
\]

\(P_{IN}\) is the input power, \(P_{OUT}\) is the output power and \(P_{dc}\) is the dc power, i.e. the power taken from the dc source.

In all PAs there is a trade off between efficiency and linearity. In order to achieve good efficiency, the PA needs to be driven at a high power level. The downside is that this introduces larger nonlinear effects in the output. A typical curve of efficiency plotted against output power is presented in Figure 2.7.

![Figure 2.7 Gain and efficiency plotted against output power.](image)

\(^3\) Taken from [4], Chapter 3.
The plot contains data derived from measurements performed on the power amplifier used in this thesis; the data file was provided by another department at FOI. It is clearly visible that the efficiency (lower curve) is strongly correlated to the power level. The nonlinear effects can be seen in the gain curve (above curve) as the gain starts to fall at higher output levels due to the compressive transfer characteristics of the PA.

An important issue concerning efficiency in PAs is the heat dissipation. It is not only of interest to achieve good efficiency due to the energy consumption; cooling is a main point. The energy fed to the circuit that is not transmitted to the output signal is converted to heat. If a large amount of energy becomes heat (low efficiency) the circuit will be warmed up, possibly producing a severe cooling problem in the system.
3 Modelling of power amplifiers

Different approaches of power amplifier modelling are mentioned in this chapter. Polynomial modelling is introduced and a thorough presentation of the chosen modelling technique, mentioning drawbacks and limitations, concludes the chapter.

3.1 Modelling techniques

Modelling of radio frequency power amplifiers (RF PA) has been a subject of intense research the last years. Not least the mobile industry has created a need for accurate mathematical models, both for evaluation of the PA itself and for synthesis of linearization and efficiency enhancing systems.

Modelling can be performed from one of at least two main starting points, which includes equivalent circuit modelling and behavioural modelling.

The equivalent circuit modelling requires deep knowledge and insight in the circuit layout, as well as analysis at component level.

The behavioural modelling approach does not involve a detailed low level description of the system. A more practical point of view is utilized by trying to describe the relationship between input and output signals in the PA. This is done by not caring about the actual cause of the signal appearance, but simply trying to represent the dynamics mathematically.

A number of different techniques exists for modelling PAs (and of course other arbitrary nonlinear systems). Mentioned below are some examples; they should not be regarded to fully cover all modelling techniques available.

Black box modelling

One approach of black box modelling can be found in [6]. The model describes nonlinear devices in the frequency domain. A describing function is used and model parameters are derived from measured data.

Neural based modelling

In the neural based modelling [7], a dynamic model is formulated. The model is trained by using input and output spectrums, providing a continuous time domain model.
**Digital predistortion of semi-linear power amplifier**

**Time series**

A new approach is found in [8]. Nonlinear time series analysis forms the fundament and a new MATLAB modelling toolbox has been developed to capture the system dynamics.

**Volterra series**

Volterra series is sometimes referred to as Taylor-series with memory\(^4\). The output signal is formed by an infinite series

\[
y(t) = \sum_{n=0}^{\infty} y_n(t)
\]

where

\[
y_n(t) = \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} \cdots \int_{-\infty}^{\infty} h_n(\tau_1, \ldots, \tau_n) x(t - \tau_1) \cdots x(t - \tau_n) \, d\tau_1 \cdots d\tau_n
\]

The functions \( h_n(\tau_1, \ldots, \tau_n) \) are called the Volterra kernels and form the basis of this model, where the output signal is an n-fold convolution of the impulse responses. However, these are difficult to derive and therefore not always practical to use in PA modelling situations. More examples of Volterra modelling approaches can be found in [4].

Of course other modelling techniques are available. Some of them focus on modelling the inner state of the PA, which is outside the scope here, or are curve fitting procedures trying to adapt mathematical expressions to measured curves. This will not always extract a model that captures the essential dynamics, or provide an analytical expression.

Despite all of the above modelling techniques, there does not seem to exist an optimal approach to modelling. This implies that the problem is complex and that further research is needed to accomplish a thorough and general approach. Including memory effects in the modelling is currently a research topic.

**Polynomial modelling**

Polynomial modelling has for long practically been the standard procedure to model the dynamics of power amplifiers. This approach may seem like the most intuitive one, since the response of PAs, when viewed in the frequency domain, clearly shows power series behaviour. The transfer characteristic is approximated with a Taylor series expansion.

\[
v_{\text{out}}(t) = \sum_{i=1}^{k} a_i v_{\text{in}}^i(t)
\]

\(^4\) The following example can be found in [3], Chapter 2.
The variable $k$ decides how long the series expansion will be, i.e. the truncation. The coefficients, $a_i$, represents the level of distortion from the individual products. If these coefficients are real-valued, only the AM/AM distortion will be modelled by the polynomial. If AM/PM distortion is to be modelled, a complex series is required; a complex representation of the input/output signal as well.

The traditional way of producing a model is to use a single input frequency and to sweep the input power level to extract data of AM/AM and AM/PM behaviour through measurements. By increasing the input level the nonlinear characteristic is examined. Then a polynomial is fitted to the measurements.

In this thesis, polynomial modelling is the approach for capturing PA dynamics. Polynomials are comprehensible and easily represented mathematically. Many other modelling approaches are more complex and demand a broad theoretical background. Furthermore, many of the above presented modelling techniques were not satisfactorily explained in papers and books, making an implementation hard to accomplish. Their model syntheses were more challenging, demanding extensive measurement setups. Therefore, polynomial modelling was used, allowing straightforward synthesis and simulation.

### 3.2 IM-based polynomial modelling

Two-tone measurements, together with power or Volterra series, have been available for long as tools for analyzing nonlinearity of power amplifiers. By applying a two-tone input signal, i.e. an amplitude modulated carrier, of the form

$$v_{in}(t) = v \cos(2\pi f_1 t) + v \cos(2\pi f_2 t)$$

the nonlinear dynamics of the amplifier may be analyzed and modelled with a polynomial,

$$v_{out}(t) = a_1 v_{in}^1(t) + a_2 v_{in}^2(t)^2 + a_3 v_{in}^3(t)^3 + ...$$

However, the derivation of the coefficients needs to be done with consideration. Firstly, the question of how many coefficients that are needed to characterize the dynamics is raised. It is not likely that a power series of order 51 will reflect the input/output relation adequately, since the higher order products are, if they exist, insignificant [5, Chapter 3]. The model order is therefore a key point in the synthesis. The dynamics, as seen in the previous chapter, is dominated by the lower order powers.

---

5 Example from [2], Chapter 7.
The exact conclusion which order to choose needs consideration. A pure “fitting” of measured data gives no practical hint since the fitting can be done accurately with different polynomial orders. In Figure 3.1, two polynomials of degree five and seven respectively are plotted. As may be noticed, the difference between the two polynomials is very small. However, dissimilar results are obtained in the frequency domain, since the seventh order polynomial introduces more frequency components. Hence, the need for a decision basis is apparent.

In [5, Chapter 3] there is a modelling procedure that takes into account the behaviour and dynamics of the PA. The two-tone setup is used as a characterization tool. By looking at the IMDs, the model order may be derived. The amplitudes of the individual frequency components are utilized for deriving the model order. In Figure 3.2, from [5, Chapter 3] there is a plot of the 3rd, 5th and 7th order IM-products. Since a model order of five not includes products of higher degree than five, it is impossible for such a model to characterize any 7th order IMs. There is a clearly visible dip in one of the IM5 magnitudes. That is a result of the 7th order IM acting on the 5th order IM, which have opposite signs (i.e. the coefficients in the polynomial have opposite signs). This implies that any dip in the actual characteristics of the IM5 cannot be sufficiently modelled with a 5th degree polynomial.
Modelling of power amplifiers

By using the IM-plots in the modelling method, a clear connection between measurements and PA dynamics is achieved. The procedure consists of 6 points, which are explained below. The algorithm is taken from [5, Chapter 3].

1. Start off with a power series and an amplitude IM plot.
2. Determine the highest order IM (“n”) which is significant for characterization purposes, at the maximum peak power level (say, -50dBc).
3. Assume the highest significant IM order has a simple n:1-dB rolloff.
4. Determine the nth degree term from IMn level at maximum drive.
5. Fit successive lower IM plots using the (now determined) higher-degree terms and a chosen value of current degree IM; just use positive and negative values.
6. Using IM phase plots (if available), obtain Volterra phase angles [i.e., from 0° or 180° values used in (5)] and refine overall fit.

In the first point, the power series modelling form is introduced. The modelling utilizes the IM-plots, which are basically plots of the magnitudes of the IM-products at different input power levels; from low levels up to highest necessary (or available) for characterization.

The highest order significant for characterization is then chosen with the plots as starting point. This is where the actual order of the power series is determined; any contribution from products of higher order than the chosen is considered to be insignificant or of less importance. The figure, -50dBc, means that these products are 50 dB lower than the peak power of the carrier frequencies and is given as an example.
Digital predistortion of semi-linear power amplifier

The assumption in point three means that no dip is present in the plot of the highest order IM. In other words, the IM is not affected by products of higher degree. Therefore, the slope of the amplitude curve will simply be dependent on the magnitude of the exponent. When plotting the amplitude curve in dB, the \( n \)th order term will have a slope of \( n:1 \).

With the help of a table (Table 3.1, presented on page 22) the coefficient may be derived for the highest power in the series. The coefficient is chosen to match the measured power levels as good as possible; the matching should be done to reflect the amplitude especially well at the highest input level (which is the most interesting level in this work).

Point five continues the work begun in the previous point. By adding the higher order contributions to the lower IM, a coefficient value can be derived. The first coefficient to start with is the one next under the highest degree. This procedure is repeated for every coefficient until all values are derived.

The sixth point requires measurements that are not always possible to make. Phase measurements involve special setups and will not be handled in this work. For this reason, only positive or negative values will be assigned to the coefficients. If the measurements had been available, the coefficients would have been complex, each with an individual phase derived from phase angle measurements, obtained at low input levels. They are referred to as Volterra phase angles [5, Chapter 3].

Model power considerations

When extracting the model coefficients, only the odd order terms are considered. The reason for this is the assumption that the even order contributions are out of band and therefore not interesting. The amplifier is assumed to have bandpass characteristic. This could either be the case if the amplifier has a narrow bandwidth or if the frequencies outside the fundamental zone can be filtered out. In our case, filtering of the high (and low) frequency components will make sure a bandpass characteristic is present. As has been stated before, the odd order powers are the worst when it comes to distortion since they are closest to the carriers.

The extraction of the (odd-order) coefficient values is done with help of some trigonometric relationships. The input signal is, as mentioned, a sum of two separate (co)sines with frequencies of some separation [5, Chapter 3]. The separation needs to be selected to make sure that the IM components are in-band, i.e. not too largely spaced. The input signal is

\[
\phi_m(t) = \nu \cos(\theta_c - \theta_m) + \nu \cos(\theta_c + \theta_m) = 2\nu \cos(\theta_m) \cos(\theta_c)
\]

where \( \theta_c \) is the carrier and \( \theta_m \) is the modulating angular frequency; the modulating frequency is chosen to be much smaller than the carrier frequency.

The odd-order power series yields\(^6\)

\( \nu \)
\( \theta_c \)
\( \theta_m \)
\[ \text{The following expressions and relations are taken from [5], Chapter 3.} \]
\[ v_{\text{out}} = a_1 2 v (\cos(\theta_m) \cos(\theta_c)) + a_3 2^3 v^3 (\cos(\theta_m) \cos(\theta_c))^3 + a_5 2^5 v^5 (\cos(\theta_m) \cos(\theta_c))^5 + \ldots + a_n 2^n v^n (\cos(\theta_m) \cos(\theta_c))^n \]

De Moivre’s theorem may be applied to the \(n\)th-degree expansion, giving

\[
\cos^n \theta = \frac{1}{2^{n-1}} \left[ \cos n \theta + n \cos(n-2) \theta + \frac{n!}{(n-k)!k!} \cos(n-2k) \theta \right], \quad 1 < k \leq (n-1)/2
\]

(N.B. \(k\) should be equal to or smaller than \((n-1)/2\), not only smaller as in [5])

By assuming a narrow band-limited system, the following expression can be derived for the \(n\)th term of the output, a modulation on the fundamental carrier

\[
v_{\text{out}} = a_n 2^n v^n \frac{n!}{2^{n-1} \left( \frac{n-1}{2} \right) \left( \frac{n+1}{2} \right)} \frac{1}{2^{n-1}} \left[ \cos n \theta_m + n \cos(n-2) \theta_m + \frac{n!}{(n-k)!k!} \cos(n-2k) \theta_m \right] \cos \theta_c
\]

The same restrictions for \(k\), \(1 < k \leq (n-1)/2\).

To exemplify, consider the third order products. The above expression yields

\[
v_{\text{co3}} = \frac{3a_3 v^3}{2} \left( \cos 3\theta_m + 3 \cos\theta_m \right) \cos \theta_c = \frac{3}{2} a_3 v^3 \left\{ \frac{1}{2} \cos(\theta_c - 3\theta_m) + \frac{1}{2} \cos(\theta_c + 3\theta_m) + \frac{3}{2} \cos(\theta_c - \theta_m) \right\}
\]

The contributions on the carrier frequencies and the third order IMs are proportional to input amplitude and coefficient values. Note that the contribution is divided by two compared to the first expression, since the power is split on the upper and lower carriers and IMs.

A complete table for coefficients up to eleventh grade can be found in Table 3.1. The table follows Table 3.1 in [5, Chapter 3] with the exceptions of adding the eleventh grade coefficients and correcting the nine-column values, which are wrongly printed in the book.
Table 3.1 Coefficient values for IM-products. IM1 refers to the carrier.

<table>
<thead>
<tr>
<th>n</th>
<th>3</th>
<th>5</th>
<th>7</th>
<th>9</th>
<th>11</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>$a_3v^3$</td>
<td>$a_5v^5$</td>
<td>$a_7v^7$</td>
<td>$a_9v^9$</td>
<td>$a_{11}v^{11}$</td>
</tr>
<tr>
<td>IM1</td>
<td>9/4</td>
<td>25/4</td>
<td>1225/4</td>
<td>3969/64</td>
<td>53361/256</td>
</tr>
<tr>
<td>IM3</td>
<td>3/4</td>
<td>25/8</td>
<td>735/64</td>
<td>2646/64</td>
<td>38115/256</td>
</tr>
<tr>
<td>IM5</td>
<td>-</td>
<td>5/8</td>
<td>245/64</td>
<td>1134/64</td>
<td>38115/512</td>
</tr>
<tr>
<td>IM7</td>
<td>-</td>
<td>-</td>
<td>35/64</td>
<td>567/128</td>
<td>12705/512</td>
</tr>
<tr>
<td>IM9</td>
<td>-</td>
<td>-</td>
<td>-</td>
<td>63/128</td>
<td>2541/512</td>
</tr>
<tr>
<td>IM11</td>
<td>-</td>
<td>-</td>
<td>-</td>
<td>-</td>
<td>231/512</td>
</tr>
</tbody>
</table>

**Drawbacks and limitations**

Polynomial modelling has, as all modelling techniques, a number of drawbacks, more or less serious.

The polynomial is static and memory-less. Static means that the coefficients are not variable, giving the same polynomial independent of frequency or bandwidth of input signal. Memory-less indicates that no influence from previous signals is present on the output. In reality, frequency, bandwidth and signal history will affect the nonlinear response. By using an input signal with a bandwidth larger than zero, some bandwidth dependent dynamics will be included in the derivation of the coefficients.

A general flaw with polynomials is the limited input level that can be handled. Any polynomial will grow towards infinity or minus infinity as the input grows. Thus, it is of importance that the input signal peak is not allowed to be too large, since the output may be significantly out of normal behaviour. There is a justification of this effect in a physical sense; a power amplifier cannot be fed with an arbitrarily large input because that will break the circuit. This must be considered when using the modelling polynomial.

As the coefficients are real-valued, no phase distortion effects are included. To compensate this, a phase alternating part is added to the model implementation. By measuring the phase response of the amplifier, this may be applied in the frequency domain with use of the FFT. The FFT provides amplitude and phase information for all frequencies (up to the Nyquist frequency) in the signal. When generating the output signal (with IM-frequencies), the result can be Fourier-transformed and an amplitude dependent phase change acting on all frequencies applied. This will somewhat compensate for the shortcoming of the polynomial to model phase change.
Some approaches to linearization of power amplifiers are briefly discussed in this chapter. The main focus is on digital predistortion and the technique examined in the thesis is presented.

### 4.1 Linearization techniques

Linearization is used to make (nonlinear) systems behave more linear. In the power amplifier case, this means less spectral distortion. It involves some kind of signal processing, before or after the PA and aims at making the cascade of subsystems working more linear than the PA itself.

There are three main techniques for linearization:

- Feedforward
- Feedback
- Predistortion

**Feedforward**

Feedforward linearization uses an additional amplifier to manipulate the output from the PA. A portion of the signal from the PA is fed into a further path, where the input signal to the PA is subtracted. What are left are the distortion components, and they are fed into the extra amplifier. The signal from this amplifier is added with anti-phase to the signal path of the main amplifier and at the output a signal with less distortion is present [9, Chapter 4].

**Feedback**

Feedback is a well-known and established way of dealing with signals in control theory. In PAs the technique may be used to improve the linearity. A number of different configurations are available, for example RF feedback, active RF feedback, distortion feedback and modulation feedback [3, Chapter 4].

### 4.2 Predistortion

The basic idea of predistortion is simple. Before feeding the signal to the PA, distort it to make the output from the PA more linear. This might seem like a contradiction; to
Digital predistortion of semi-linear power amplifier

improve the distortion, introduce additional distortion. But if the correct distortion is added, the result will be improved output.

A basic system description is depicted in Figure 4.1.

The goal of the predistortion system is to make the cascade of the predistorter (PD) and power amplifier linear. This is achieved by designing the PD to be as close as possible to the inverse of the PA function.

If the transfer functions of the two systems are \( F(x) \) and \( G(y) \), the cascaded system response will be

\[
z = G(y) = G(F(x))
\]

When \( G(F(x)) \approx Kx \), a linear response has been achieved by letting two nonlinear systems work together.

The problem is naturally to find a proper inverse to the PA function. A variety of techniques exist, which are more or less complicated and able to produce satisfying results. As an example, cubic predistorters can be mentioned. They focus on suppressing the 3\(^{rd}\) order IM-products, but are not able to cope with higher orders of distortion.

The fast development of digital circuits has made digital predistortion a subject of intense research. However, the analogue approach will also be mentioned.

4.2.1 Analogue predistortion

A couple of analogue techniques can be brought up. One of the simplest PDs is a nonlinear element consisting of a diode in front of the PA. The diode is nonlinear in behaviour and may therefore be used for this purpose. Another possibility is to use an additional amplifier, similar to the main amplifier. By using the distorted signal from the supplementary PA and adding it to the main signal path with a negative sign, a predistorted input is introduced to the main PA [10].
4.2.2 Digital predistortion

The interest in digital predistortion has been growing, commercially and scientifically, the recent years [11]. Thanks to the rapid digital electronics development, more advanced and complex solutions have been made possible to implement.

Digital predistortion can be implemented, for example, with the use of look-up tables in a DSP (Digital Signal Processing) system. The RF signal is measured and a memory is addressed; the address depending on signal amplitude (and phase). The memory contains information of correction to the signal in order to achieve a suitable, predistorted signal as input to the power amplifier. A measurement of the power amplifier characteristics must be done in advance, in order to apply the right correcting signal updating. The approach can be made adaptive, which means that the addressed memory is updated. The predistorter will then be able to adapt to changes in transfer characteristics caused by for example varying ambient conditions.

The digital predistortion approach is often said to have a bright future in linearization of PAs. However, the technique suffers from some flaws. Perhaps the most serious one is that many algorithms are unable to compensate for memory effects in the PA. Handling these effects in PD systems is currently an area of research in the PD field.

4.2.3 Digital PD in frequency domain

The approach in this thesis forms a new way of handling digital predistortion. The technique does neither rely on models of the amplifier, nor does it require large look-up tables to synthesize the predistortion signal. Instead of starting out in the time domain, a frequency domain viewpoint is used. This can be seen as a time domain predistortion with a change of basis. Phase shift in the frequency domain corresponds to letting the signals move around in time in relation to each other.

The basic jamming system setup is given in Figure 4.2.

![Figure 4.2 Basic system description; signal processing unit, power amplifier, A/D and D/A conversion units and antenna. PA signal is fed back to SP. Input to SP selects which frequencies to jam.](image_url)

The SP (signal processor) contains all signal-processing equipment. Frequencies selected for jamming are given as input to the system ($\Sigma f$). The predistortion idea is to let a portion of the output signal from the PA be fed back to the system. The analogue RF signal is converted to digital form by an ADC (Analogue to Digital Converter). By processing the (digital) signal in the frequency domain, with Fast Fourier Transforming,
information of the frequency content of the signal is available. Using this information, a predistortion signal is synthesized. This signal is converted from digital form to analogue and then processed in the PA; the output is transmitted and also fed back to perform another loop in the PD system. Hence, the technique is dependent on real-time measurements of the output signal from the PA and computationally fast circuits in the signal processing unit.

To suppress the unwanted frequency components, which is the overall goal with all linearization techniques, the PD-signal is created consisting of the wanted components (carriers) and the unwanted signals in anti-phase. Anti-phase means a $180^\circ$ phase shift, the same thing as adding a minus sign to a (co)sine signal. By letting this new, “inverse”, signal be processed in the amplifier, compensation for the unwanted frequencies is obtained.

In Figure 4.3, a graphical description of the basic idea is given. The solid lines represent the uncompensated output spectrum, which also includes the dotted carriers. The predistortion signal is composed of the dotted lines (including the carriers as well). After processing this new signal, differing from the original input with added (predistortion) frequency components, suppression of the unwanted frequencies is achieved.

To acquire such a result, the signals (i.e. the frequency components) need not only be perfectly phase matched, but also equal in amplitude. What happens when the compensating signals not are perfectly matched to the nonlinear response can be seen in Figure 4.4. Adding the signals basically means summing vectors. If they are not anti-phase, the result will be a signal of some amplitude, depending how much the phase differs from ideal. The same is of course valid for the amplitude.
A possibly bigger problem with the addition of extra frequency components comes from the nonlinear nature of the amplifier. When introducing more signals, new frequency products will appear at the output, resulting from the sum and difference products formed in the PA. Therefore, an iterative algorithm will be necessary in order to provide compensation for the new unwanted frequencies. By letting the algorithm iterate a number of times, an improved spectrum will hopefully result. Another concern with adding more energy is the enhanced power fed to the amplifier. If it is driven hard, near saturation, it may perhaps not be worthwhile to add more signals and enhance the amplitude. However, the signals added have much lower amplitude than the carrier signals, which should mean that no big problem occurs when adding more components.

This type of predistortion has never been used before, despite its apparently simple approach. The reason for that is that the technique for digitizing the signals very fast has not been available before, especially not the synthesis of the signals. The SP-system must be able to produce many signals of various frequencies and phases at the same time.
5 Measurements

The setup used for measuring amplifier distortion levels is presented together with considerations of the measurement procedure. Frequently used units and useful conversions are described.

5.1 Measurement setup

To be able to synthesize the model, measurements needed to be performed. The magnitudes of the individual (IM) frequency components had to be determined at different input power levels. For this purpose, a measurement setup was produced. The setup consisted of two signal generators, one pre-amplifier, two combiners/dividers (T), dampers (D) and a computer with two channels for data acquisition. In Figure 5.1 the setup is presented.

![Figure 5.1 Measurement setup.](image)

The signal generators were one hp 8663A Synthesized signal generator and one Rhode & Schwarz Signal generator 819.0010.52. The pre-amplifier was a Mini Circuits LZY-1. Combiner/dividers are used to add two signals or divide a single signal. A number of different dampers were used with the purpose of decreasing the signal level.

5.1.1 Considerations

It was decided that the amplifier should be characterised in a band of 70 MHz, chosen to be 110-180 MHz. This because the communication in the military band is between 30 and 90 MHz, which means a total bandwidth of 60 MHz. The power amplifier, on the other hand, has a bandwidth stretching from 100 MHz to 500 MHz. In the real system implementation, a frequency (up)conversion needs to be done in order to use the power
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amplifier. In this work, no up- or downconversion is considered, while that is not necessary for testing of the predistortion algorithm.

However, the sample rate of the computer is only 204.8 MHz. If signals in the 110-180 MHz interval are sampled, the measurements will inevitably suffer from aliasing. Fortunately, this does not need to constitute a big problem. Since the carrier frequencies of the input signals are known (being in the 110-180 MHz band), one can calculate the alias frequencies for these and also for the interesting in-band IM-products. Thereby, the undersampling is only a matter of calculating the real frequency values when the measurements are presented in a plot. The frequency axis will span from 0 Hz to the Nyquist frequency (102.4 MHz) in the FFT plot.

The possibility to measure two channels simultaneously makes it possible to achieve data of the PA phase characteristics. By comparing the phase of the signal from the PA to the phase of the signal that is not amplified, an insight of the PA phase characteristics may be derived. This is achieved by comparing the different signals in the two signal paths in Figure 5.1. Note that only the phase of the carriers may be compared. There is no possibility to view changes of the IM-phases since no input signals of the same frequencies are available.

In order to make the phase calculations, it was necessary to calibrate the measurement setup. Because of delays in the cables and dampers which connect the different parts, signal phases will be altered (i.e. time delays are introduced). To compensate for that a measurement was done with only the cables and dampers connected to the computer. When comparing the phase difference between the two channels, a value of the delay between the two signal paths can be obtained. The dampers are necessary because the input levels would be far too high for the computer hardware without them.

Another issue is the pre-amplifier. The pre-amplifier is vital because it gives the necessary boost for the signals from the generators before being fed to the PA. The signal generators themselves are not able to produce large enough amplitudes. But since the pre-amplifier is a PA itself it suffers from nonlinear behaviour as well. Consequently, the input signal to the PA (device under test) includes IM-products. These needed to be compensated for if they are considered to affect the measurements. To have some control of the IM-levels, the pre-amplifier was used with the same input level for all different input levels to the device under tests, i.e. the signal generators delivered the same power level all the time. Varying damper configurations were used to enable alternating power levels to the two signal paths. These dampers were inserted between the pre-amplifier and the divider (in Figure 5.1 labelled D).

On the computer, a MATLAB environment was used to present the measurements. The signals from the two channels were recorded using special hard- and software and the result was read into MATLAB for further analysis. The fast Fourier transform, FFT, in MATLAB enabled amplitude and phase data to be collected for the varying frequencies.
5.2 Power units and conversions

Amplitudes are often given in the unit of Volt (V). Radio technology deals with signals in a wide span, with powers that can be 100 W from a transmitter down to fractions of mW in receivers. This raises the need for a suitable unit when dealing with signal powers. A standard unit frequently used is dBm,

$$P_{\text{dBm}} = 10\log_{10}(P_{\text{mW}}/1\text{mW})$$

where $P_{\text{dBm}}$ is the power in dBm and $P_{\text{mW}}$ is the power in mW.

Spectral analyzers are measuring devices which can present the input signals in the frequency domain, giving the frequency and power; often in dBm. To convert the power in dBm to voltage amplitude, the following expression is used

$$v = \sqrt{\frac{10^{\left(P_{\text{dBm}}/10\right)}}{1000}} R_{\Omega}$$

where $v$ is in volts and $R_{\Omega}$ is the resistance.

In many systems and in the systems covered in this thesis $R_{\Omega}$ equals to 50 ohms, a standard value. The voltage calculation yields the rms-value (root mean square),

$$v_{\text{rms}}^2 = \frac{1}{T} \int_{0}^{T} v_s^2(t)dt$$

where $v_s$ is the voltage signal. $v_{\text{rms}}$ equals the dc voltage that would produce the same heating as one cycle of the sine wave; the peak amplitude of the corresponding sine wave is $\sqrt{2}$ times the $v_{\text{rms}}$ [9, Chapter 1].

Another unit that can be found in the literature is the dBc, where the reference is a carrier wave. If, for example, a signal is said to be -50 dBc compared to a reference of 40 dBm, the magnitude level is -10 dBm; i.e. the level is 50 dBm lower. dBc is sometimes used when relating IMs to the carrier amplitudes, as a measure of the nonlinearity.
Digital predistortion of semi-linear power amplifier
6 Modelling and simulations

This chapter presents and summarizes the results obtained in the work. First, the modelling of the power amplifier is presented and then the outcome of the predistortion simulations is given.

6.1 Modelling

To produce the polynomial model, measurements were performed to extract magnitudes of the different IM-products. This also provided a decision basis for selecting an appropriate modelling order. An amplifier recently developed and intended to be used in the real system, was chosen for modelling. Since the amplifier recently was constructed, no data sheets were available. The only data obtainable were measurements of the third order IMs, figures of the PA gain and PAE figures collected at five different frequencies. A data sheet of the phase (time) delay was also available.

Measurements were carried out with the setup described in Chapter 5, Measurements. Biasing (gate and source voltages of the PA) was kept at a constant level. Input signal power levels were chosen not to exceed the recommended maximum input level. The power amplifier has an approximate maximum output level of 10 W, which equals to 40 dBm. In this case, a total input power of 23 dBm was considered to allow modelling without risking breaking the circuit. This input level gives an output level of about 37 dBm. In Table 6.1 values from the data acquisition is presented. -100 dBm indicates that no value was obtained, due to low power level. The measuring equipment had a noise level of about -80 dBm, levels below were not possible to detect. Note that the values in the table not are what were detected in the computer since the signals coming in on the two channels were damped in order to avoid excessive signal levels; this has been compensated for.

After conversion to voltage unit (from dBm), modelling was carried out. Two polynomials were constructed by using the procedure described in Chapter 3.2, IM-based polynomial modelling; one of 11th and one of 9th degree, presented below.

\[
v_9 = 5.2v + 1.1 \cdot 10^{-1}v^3 - 1.29 \cdot 10^{-2}v^5 + 4.3 \cdot 10^{-4}v^7 - 5 \cdot 10^{-6}v^9
\]
\[
v_{11} = 4.2v - 4 \cdot 10^{-3}v^3 + 3.05 \cdot 10^{-3}v^5 + 1.4 \cdot 10^{-4}v^7 - 1.15 \cdot 10^{-5}v^9 + 1.7 \cdot 10^{-7}v^{11}
\]
Digital predistortion of semi-linear power amplifier

Some testing was carried out to assure that the models performed acceptable in relation to the measured values. Both models produced satisfying results, the ninth order model somewhat better at low input levels. No extensive testing was carried out with different input signals and power levels, since the modelling was not the main focus in the work. The main purpose of the synthesis was to extract models to be able to carry out the predistortion simulations. Two models were synthesized with the chosen modelling algorithm, because it was of interest to test whether there would be any difference in the simulations. No essential differences were noticed between the two polynomials when simulating the predistortion algorithm.

Table 6.1 Measured magnitudes of distortion products

<table>
<thead>
<tr>
<th>Name</th>
<th>Carr. 1</th>
<th>Carr. 2</th>
<th>IMD 3 low</th>
<th>IMD 3 high</th>
<th>IMD 5 low</th>
<th>IMD 5 high</th>
</tr>
</thead>
<tbody>
<tr>
<td>Frequency MHz</td>
<td>144</td>
<td>146</td>
<td>142</td>
<td>148</td>
<td>140</td>
<td>150</td>
</tr>
<tr>
<td>Input level dBm</td>
<td>1</td>
<td>16,2</td>
<td>16,1</td>
<td>-32,5</td>
<td>-32,6</td>
<td>-48,8</td>
</tr>
<tr>
<td></td>
<td>6</td>
<td>20,5</td>
<td>20,4</td>
<td>-22,2</td>
<td>-22,4</td>
<td>-45</td>
</tr>
<tr>
<td></td>
<td>9</td>
<td>23,6</td>
<td>23,5</td>
<td>-14,1</td>
<td>-14,2</td>
<td>-42,5</td>
</tr>
<tr>
<td></td>
<td>11</td>
<td>26,2</td>
<td>26,0</td>
<td>-9,2</td>
<td>-9,7</td>
<td>-34,2</td>
</tr>
<tr>
<td></td>
<td>16</td>
<td>30,1</td>
<td>29,9</td>
<td>-3,3</td>
<td>-3,3</td>
<td>-13,4</td>
</tr>
<tr>
<td></td>
<td>18</td>
<td>32</td>
<td>31,8</td>
<td>1,3</td>
<td>1,4</td>
<td>-15,4</td>
</tr>
<tr>
<td></td>
<td>19</td>
<td>32,6</td>
<td>32,5</td>
<td>5,7</td>
<td>5,8</td>
<td>-10,6</td>
</tr>
<tr>
<td></td>
<td>20</td>
<td>33,4</td>
<td>33,3</td>
<td>9,8</td>
<td>9,6</td>
<td>-6,6</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>IMD 7 low</th>
<th>IMD 7 high</th>
<th>IMD 9 low</th>
<th>IMD 9 high</th>
<th>IMD 11 low</th>
<th>IMD 11 high</th>
</tr>
</thead>
<tbody>
<tr>
<td>138</td>
<td>152</td>
<td>136</td>
<td>154</td>
<td>134</td>
<td>156</td>
</tr>
<tr>
<td>1</td>
<td>-51</td>
<td>-51</td>
<td>-100</td>
<td>-100</td>
<td>-100</td>
</tr>
<tr>
<td>6</td>
<td>-43,9</td>
<td>-42,5</td>
<td>-100</td>
<td>-100</td>
<td>-100</td>
</tr>
<tr>
<td>9</td>
<td>-44,9</td>
<td>-43,7</td>
<td>-100</td>
<td>-100</td>
<td>-100</td>
</tr>
<tr>
<td>11</td>
<td>-38,1</td>
<td>-38,8</td>
<td>-100</td>
<td>-100</td>
<td>-100</td>
</tr>
<tr>
<td>16</td>
<td>-26</td>
<td>-25,5</td>
<td>-100</td>
<td>-100</td>
<td>-100</td>
</tr>
<tr>
<td>18</td>
<td>-12,6</td>
<td>-14,1</td>
<td>-24,9</td>
<td>-25,1</td>
<td>-100</td>
</tr>
<tr>
<td>19</td>
<td>-12,8</td>
<td>-13,3</td>
<td>-16,3</td>
<td>-16,9</td>
<td>-100</td>
</tr>
<tr>
<td>20</td>
<td>-16,9</td>
<td>-19,2</td>
<td>-11,8</td>
<td>-11,8</td>
<td>-22</td>
</tr>
</tbody>
</table>

It was also desirable to have some kind of modelling of the phase behaviour of the amplifier. To meet this requirement, the phase shift for the carriers at the various power levels was obtained. In Figure 6.1, carrier phase characteristics are plotted. Reference phase is the value acquired from the lowest power input value. A phase shift was introduced in the model, acting on the input frequencies and being proportional to the amplitude. Thereby some kind of AM/PM effect is handled in the model, as the coefficients are real-valued and hence do not provide phase alternation. The phase shift is also plotted in Figure 6.1, selected to be near the mean of the carrier shifts. The reason for choosing this phase shift was that the carriers had not exactly the same shift. The selected
phase shift will still be near the measured; in the plot, it can be seen that it is not very large (about 8 degrees at the highest input level).

![Figure 6.1 Measured phases for carriers, mean and model phase.](image)

The power amplifier will inevitably introduce a delay in the signal path when signals are processed. A time delay can be seen as a phase shift for sinusoidal signals. Hence, in order to model the time delay in the PA, a phase shift proportional to the frequency was introduced. The phase shift (i.e. time delay) was measured and compared to a data sheet provided by FOI, which contained measurements performed at another department. Some discrepancy was found, but the overall resemblance proved to be adequate. Phase delay for the modelling can be seen in Figure 6.2 together with measured phase and data sheet values. The phase shift was measured for frequencies between 110 and 180 MHz with 3 MHz spaces.

![Figure 6.2 Phase delay from data sheet, measured and fitted.](image)
6.2 Predistortion results

The implementation of the predistortion algorithm followed the basic idea described in Chapter 4.3.2, Digital predistortion in frequency domain. To achieve an algorithm that provides improved spectral properties, a couple of variables needed to be derived.

- **Scaling**  
  The compensating signal needs to be scaled with appropriate magnitude; otherwise the amplifier may be fed with excessive power levels or too low signal amplitudes.

- **Phase delay**  
  Since the PA will delay all signals, this needs to be compensated for in order to achieve correct phases.

When having developed a functioning algorithm, testing of the predistortion performance was carried out. A number of different questions concerning the behaviour of the algorithm were examined.

- **Distortion improvements**  
  The main point is of course to test whether any improvements can be achieved and, if that is the case, how big they are.

- **Varying signals**  
  By varying the number of input signals and their amplitudes, a test of the predistortion performance with different signal constellations is obtained.

- **Compensation of arbitrary frequency bands**  
  Sometimes improvements in a small frequency band are wanted. This should be examined.

- **Iteration stop**  
  It is of interest to test whether a decision for iteration stop may be implemented. If no improvement in the output signal is evident after iterating a number of times, no further updating of the predistortion signal is necessary.

- **Frequency switch**  
  When switching frequency/frequencies at the input, new unwanted signals are produced. The question is whether the old PD spectrum should be updated or if the result of the algorithm is better when resetting the PD signal.

- **Phase alternation**  
  An interesting thing to observe is how the algorithm performs when phase is altered in the signals, i.e. nonideal phases in the PD signal.
6.2.1 Considerations

The signals in the system are sampled and converted to digital form in an ADC (analogue to digital converter) at the input. The reverse operation (DAC) is performed at the output. This will introduce quantization distortion, both at output and input. To account for this effect, the signals are quantized both before and after processing them in the PA model. The input ADC is assumed to have a 12-bit conversion, spanning 1.5 Vpp (Volt peak-to-peak). The output DAC is assumed to have 10-bit conversion, spanning 1 Vpp. This results in quantization levels of

\[ \frac{1.5}{2^{12}} \approx 3.66 \cdot 10^{-4} \text{ Volt/level for the ADC and } \frac{1}{2^{10}} \approx 9.77 \cdot 10^{-4} \text{ Volt/level for the DAC.} \]

The basic algorithm for the predistortion was implemented utilizing the MATLAB \texttt{FFT} function for calculation of the compensating signal. By reversing the phase of the frequency components given by the \texttt{FFT}, a frequency domain predistortion signal was achieved. The phases for the carrier signals were preserved. Inverse transforming then yielded a real-valued signal in the time domain.

6.2.2 Derivation of variables and simulation properties

Firstly, the basic algorithm was implemented, i.e. a simple 180 degree phase alternation of the input signals, with the exception of the carriers which were added with wanted phase and amplitude. To derive an appropriate scaling, testing was carried out with different magnitudes. Intuitively, a scaling close to the PA amplification should be appropriate; scaling in this case means dividing the signal amplitude with a scalar value. This proved to be correct during the testing. A scaling lower than the amplification makes the input signals too large. On the other hand, a large scaling makes the algorithm slow in the sense that it takes more time (iterations) to obtain a good outcome of the predistortion. In the actual implementation, the magnitude of the feedback signal must be known in order to scale correctly. In the simulations, the entire signal was fed back in the loop, but in real conditions a portion will be handled in the predistortion algorithm. Hence, amplification will be necessary in order to achieve a suitable signal level.

The phase compensation was implemented to match the specifications provided by the measurements. Since the measured phase (time) delay is modelled in the PA model, a perfect compensation may be used. However, to capture discrepancies in the simulations, a phase deviation of +10 degrees was introduced. A perfect phase match for handling time delay would probably not be possible to obtain in the real system.

Memory and thermal effects are not handled in the model. To include some dynamics for nonideal signal phase in the simulations due to these effects, a randomized phase alternation was implemented in the model. A normally distributed variable with mean 0 and variance 1 was multiplied with 40 and used for deciding the phase shift for all individual frequencies. All frequencies were phase alternated with some tens of degrees at most. The phases were randomly chosen every time the signal was processed in the PA model.
The above derived variables were used in the simulations presented below. The frequency band considered was 110-180 MHz, all other frequencies are assumed to be filtered out. All signals had frequencies in this band; they were synthesized with a sample frequency of 819.2 MHz to enable a good resolution in the frequency band. Amplitudes were chosen to enable simulation stability. All results obtained are valid for the models described in Chapter 6.1, Modelling results. No significant differences were noticed between them when simulating.

6.2.3 Frequency compensation

Testing was carried out with different signal constellations. In Figure 1 in Appendix one such constellation is presented. The input signal consists of two carriers at 144 and 146 MHz. When looking at the uncompensated output signal, an amplitude difference of approximately 25 dBC can be seen between the carriers and the strongest IMD. After iterating a number of times, this value increases (Figure 2 to 4). 5 iterations yields a 20 dBC improvement, 10 iterations 10 dBC more and 20 a further 10 dBC. This is a very good result; the suppression of the unwanted components is strong.

Different frequencies were tested as input signals. Good results were obtained in all cases, especially when the carriers are located far apart in the frequency band. This means that not many of the IMDs are located inside the band of interest, which makes the number of iterations needed for good suppression lower. The reason is probably that less frequencies and energy is added to the PD signal than when many unwanted components are present. It was also observed that the simulations sometimes were unstable, yielding a signal which grew towards infinity. The reason for that is the inability of the model to handle large input amplitudes. When the saturation level is overridden in the model due to high amplitudes, an unstable behaviour of the simulation results. There is a possibility that this could be the case in real system testing; when compensating many signals and reaching saturation the strongly nonlinear output may not be possible to predistort. This must be considered when implementing the algorithm.

6.2.4 Multiple signals

The behaviour of the algorithm when handling multiple input signals was examined. 5, 10, 15 and 20 signals of different frequencies were added and used in the simulations. In all cases, suppression of the IMDs was achieved. Since large amplitudes cannot be handled by the PA model, only signals with lower powers could be examined. But the results points toward the conclusion that the algorithm will work for multiple input frequencies, provided that no excessive power levels are introduced.

No decrease in performance was seen when allowing individual amplitudes for all input frequencies. Amplitudes of course need to be considered according to the observed amplitude restrictions. An example of 5 input signals with different amplitudes is presented in Appendix. Figure 5 shows the uncompensated signal and Figure 6 PA output after 10 iterations.
6.2.5 Arbitrary frequency bands

The algorithm was tested when compensating arbitrary frequency bands. This enables disturbing signals in selected parts of the spectrum to be suppressed; a new feature not implemented in existing linearization techniques. Good results were obtained with varying number of input signals and frequency bands. In Figure 7 in Appendix an example is presented with 5 input signals and correction in two frequency bands; 120-135 MHz and 160-170 MHz. The compensated bands in Figure 8 clearly contain unwanted frequencies of lower amplitudes than the rest of the spectrum. The two carriers inside these bands are conserved and not affected by the algorithm.

6.2.6 Iteration stop

If possible, it would be of interest to implement a test in the algorithm which decides whether to continue the updating (of the PD signal) or not. When no considerable improvement can be measured at the output, it is unnecessary to continue (the updating). During the testing it was seen that a number of iterations were needed to derive a good PD signal. Somewhere around seven iterations was required in many cases; naturally differing depending on signal constellation. The test implemented evaluated the highest amplitude of the IMDs. If this amplitude was not improved (suppressed) in the next iteration step, the algorithm stopped; testing starting after the seventh iteration. A version with the mean magnitude instead of the peak amplitude was also tested. The results were ok, but it is questionable whether these algorithms are fast enough for the real implementation. More efficient and faster algorithms may be needed in the actual system.

6.2.7 Frequency switch

In real jamming situations, a number of frequencies may be jammed at the same time. These frequencies can change independent of each other and time. Therefore it is of interest to examine how the algorithm handles frequency switches. When frequencies are changed, the predistortion spectrum utilized is not likely to be ideal for the new signal constellation. The question is whether to update the old spectrum or begin compensating with a new, i.e. start algorithm from scratch. Testing implies that the spectrum should be reset and a new derived. IMD levels produced by the old PD spectrum are almost equal in magnitude compared with uncompensated products. Thus, it is probably better to restart algorithm updating. Furthermore, when considering that the memory effects not are fully covered in the simulations, this conclusion is additionally justified. Changing the signal bandwidth will probably affect the PA characteristics, making the old PD spectrum deviate further from ideal. However, testing this in the real system is recommended.

6.2.8 Phase alternation

An interesting behaviour to monitor is the sensitivity to nonideal phase compensation. That is, the compensation for delay in the PA (and generally in all other circuits). The algorithm appears to be stable for deviations of up to some tens of degrees. But too large phase errors make the output signal unstable and the PD performance is considerably degraded. This implies that an important point in the system synthesis is to calibrate the circuitry. All delays in the different signal paths must be measured in order to be able to make a PD signal as close to ideal as possible. Any phase error introduced will most likely decrease the ability to linearize the PA output.
7 Conclusions & Further work

This chapter contains the conclusions made in the study. Suggestions for further work are also presented.

7.1 Conclusions
The simulations carried out in the study imply that the new technique of digital predistortion is promising. Good results were obtained during the testing. The ability to suppress unwanted frequency components is satisfying. Interesting new features such as concentrating predistortion on selected frequency bands also proved to be possible with positive results. However, testing the technique with hardware needs to be done to confirm and measure how good results are possible to achieve in reality. The model used in the testing is not true in the sense that it captures all nonlinear dynamics. Memory effects are for example not covered at all by the model. Therefore, we are careful with statements of how well this approach will turn out to be in reality. Nevertheless, the technique has proved to have potential and the conclusion must be that it should be implemented and evaluated. The shortcomings of the modelling may not be vital since the signals are measured in real time and the updating of the predistortion algorithm is dynamic. Memory effects, uneven IM components and other unwanted phenomena are measured continuously and may be compensated for. Therefore I believe that the technique has a future, the final result naturally depending on how well the implementation turn out.

7.2 Further work
The main purpose of a further work is hardware implementation and testing the algorithm in real conditions.

Assuming that a model with more extensive dynamics would be synthesized, it could be rewarding to test the predistortion algorithm in new simulations to see if the results are comparable with those obtained in this study.

The addition of deciding iteration stop in the algorithm is a possible future piece of work. A more efficient iteration stop algorithm, i.e. one using few calculation steps would be desirable. This is probably more easily achievable when it is possible to monitor the actual behaviour of the system. It could also turn out that an iteration stop algorithm not is necessary in the real system, depending on how the algorithm behaves, the possibilities
of fast calculation and the number of iterations used before changing the signal constellation. The algorithm could for example base the stop decision on the magnitude of the highest IMD or how much the largest IMD has been suppressed since the start of the iteration.

The error measure sensitivity, in terms of for example stability, is an important feature to cover when testing the true system setup. Also, badly calibrated circuitry can be covered in such investigations.
Bibliography


Figure 1. Uncompensated spectrum.

Figure 2. Spectrum after 5 iterations.
Figure 3. Spectrum after 10 iterations.

Figure 4. Spectrum after 20 iterations.
Figure 5. Uncompensated spectrum with 5 carriers.

Figure 6. Compensated spectrum after 10 iterations.
Figure 7. Uncompensated spectrum with 5 carriers.

Figure 8. Compensated spectrum after 10 iterations.
På svenska

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