

# A new adaptive baseband pre-distortion algorithm for linearization of power amplifiers, application to EDGE-GSM transmitters

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**Abstract:** - This paper presents a new adaptive pre-distortion algorithm for linearization of Power Amplifiers and its application to EDGE-GSM transmitters. The pre-distortion system is polynomial. The criterion is minimisation of EVM (Error Vector Magnitude) evaluated on equalised signal. The analytic expression of the gradient of the criterion has been calculated. A stochastic gradient algorithm is applied using this analytic expression. The method has been tested on a class AB power amplifier with a baseband signal corresponding to EDGE-GSM Standard with  $3\pi/8$  8-PSK modulation. A special equaliser has been designed for EDGE signals.

**Key-Words:** - Linearisation, power amplifier, adaptive predistortion, EDGE, Equaliser.

## 1 Introduction

The design of very efficient power amplifiers (PA) is a crucial problem for mobile communications since it directly influences the autonomy of the mobile terminals. But efficient PA always present non-linearities that generate amplitude and phase distortions on the output signal. These distortions are characterised by AM-AM and AM-PM curves. For digital mobile communications equipment's, they are the origin of spectral regrowth in adjacent channels and of deformation of the constellation (increasing of the EVM) increasing the demodulation error rate. These distortions depend on the envelope of the modulation. For constant envelope modulations such as GMSK, they are less disturbing as for QAM modulations such as  $3\pi/8$  8-PSK of EDGE or filtered QPSK of UMTS.

Many techniques [1] have been proposed to compensate for these non-linearities such as feedback or feedforward analog techniques or adaptive pre-distortion techniques. We propose a new baseband pre-distortion technique.

## 2 Principle of the proposed method

Let us call  $z(t)$  the complex envelope of the modulated signal and  $z_I(t)$ ,  $z_Q(t)$  the cartesian co-ordinates of the complex baseband signal  $z(t)$ . This signal is supposed to be a QAM type baseband signal.

The pre-distortion function is called  $f$ . It is a complex polynomial function [2] of  $|z(t)|^2$ . For an input

$z(t)$ , the output of the pre-distortion system is  $z(t)f(|z(t)|^2)$ . With  $a=|z(t)|^2$ , the expression of  $f$  is:

$$f(a) = \sum_{k=0}^K f_k a^k. \quad (1)$$

The complex coefficients  $f_k$  are modified adaptively in order to minimise a criterion  $J$ . The proposed criterion  $J$  is an EVM criterion, it is the average value of the squared difference between the theoretical constellation points  $C_{z_e}(nT_S)$  and the "equalised" demodulated output of the amplifier  $z_{p\_a\_e}(nT_S)$ .

$$J = \sum_n e(nT_S)^2 = \sum_n (z_{p\_a\_e}(nT_S) - C_{z_e}(nT_S))^2$$

Where  $C$  is the gain of the amplifier for small power signals.

Figure 1 represents the principle of the adaptive baseband pre-distortion method.

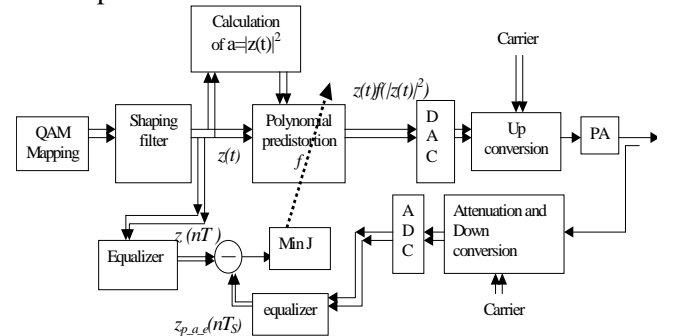


Fig 1: Diagram of the pre-distortion system

The simulations have been done using an equivalent baseband model that supposes that the output distorted

signal is narrow-band compared to the carrier frequency.

The amplifier has also been simulated in baseband. The complex gain of the amplifier is a function of the squared envelope of its input signal  $a$ . Its value is deduced from the AM-AM and AM-PM measured curves. It has been approximated by a polynomial function  $g$  of  $a$ .

### 3 Equaliser for the shaping filter of EDGE

Let us recall the main steps of an EDGE transmitter:

8-PSK mapping: the modulating bits are mapped in groups of three bits to 8-PSK symbols  $S_n$  which belongs to the mapping set  $\{e^{j2\pi k/8}$  with  $k=0,1,\dots,7\}$

Symbol rotation: at each symbol clock we turn  $S_n$  into  $S'_n=S_n e^{j3\pi n/8}$ . The new constellation is now  $\{e^{j2\pi k/16}$  with  $k=0,1,\dots,15\}$ . By this shift we prevent the complex envelope from passing through zero. (See figure 3)

Shaping filter: the modulating symbols  $S'_n$  represented by Dirac pulses excite a shaping filter whose impulse response is the GMSK pulse  $C_0(t)$  (the main component in a Laurent decomposition of the GMSK modulation).  $C_0(t)$  has a “bell” shape and covers a duration of  $5 \cdot T_s$ . After this filter the EDGE signal thus exhibits a strong level of Inter Symbol Interference and samples of this signal at the symbol rate are spread in very wide area.

If we want to measure the EVM on “clear”  $S'_n$  constellation to adapt our pre-distortion we need a filter, which equalises this  $C_0(t)$ . We can determine this equaliser by the frequency following approach.

We are expecting that the succession of the  $C_0(t)$  filter and its equaliser  $h(t)$  produce no ISI. We then choose  $H(f)=TF(h(t)) = \text{Nyquist}(f)/C_0(f)$  for  $f < F_s$  and  $H(f) = \text{Nyquist}(f)$  beyond. The trick for practical implementation is to control the delays introduced by FIR filters and to choose carefully the roll-off  $\alpha$  of the Nyquist filter so as to minimise the resonance of the subsequent equaliser.

The following figure gives the transfer functions implied.

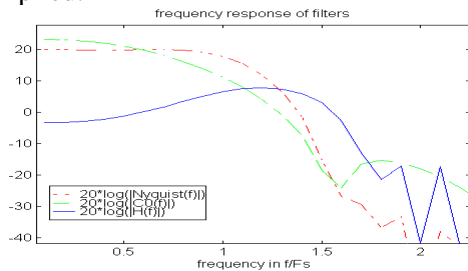


Fig 2: frequency response for Nyquist filter, shaping

filter and Equaliser filter

After this equaliser, the received signal sampled at  $F_s$  frequency exhibits the right mapping. The following figure shows the constellations obtained before and after the equaliser.

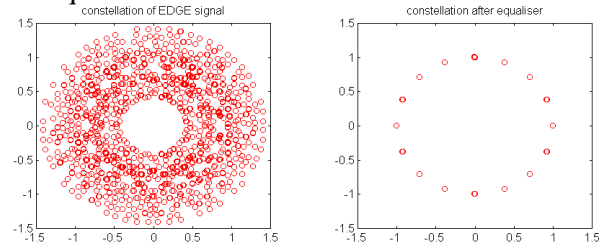


Fig 3: Edge signal and equalised Edge signal Constellations.

### 4 LMS algorithm for pre-distortion adaptation

According to the diagram of figure 1 and the choice described in 1.2, we are given the following instantaneous criterion to minimise for adapting the pre-distorter (order  $p$ ):

$$EVM(n) = |z_{p-a-e}(n) - C z_e(n)|^2$$

We have tested the simplest LMS algorithm on that criterion, that is:

$$\vec{f}^{(n)} = \vec{f}^{(n-1)} - \mu \text{grad}_{\vec{f}}(EVM(n))$$

The gradient of  $EVM(n)$  needs some steps of calculation. After these steps we obtain for the  $k^{\text{th}}$  coefficient  $f_k$  of the distorter:

$$\frac{\partial EVM(n)}{\partial f_k} = 2 \text{Re} \left\{ \left( \sum_i h_i z_{n-i} (f(a_{n-i}) g(b_{n-i}) - C) \right) \times \left( \sum_i h_i z_{n-i} a_{n-i}^{p+1-k} (g(b_{n-i}) + b_{n-i} g'(b_{n-i})) \right)^* \right\}$$

where  $a_{n-i}$  holds for  $|z_{n-i}|^2$  and  $b_{n-i}$  holds for  $a_{n-i} |f(a_{n-i})|^2$ .

Interpretation

We observe that the correction increases with :

1.  $f(a_{n-i})g(b_{n-i})-C$  which represents the error in terms of gain.
2.  $a_{n-i}^{p+1-k}$  which is the partial derivative  $\partial f(a)/\partial f_k$
3.  $g(b_{n-i})+b_{n-i}g'(b_{n-i})$  which is dependent on the local gain and local slope of the gain of the amplifier

### 5 Experimental results

We have made some simulations of our algorithm for an amplifier of class AB (weakly non-linear) developed in our laboratory [3]. As we work fully in base-band for the moment, we have considered an approximation of the gain characteristics of that amplifier by a 12-order

polynomial (figure 4).

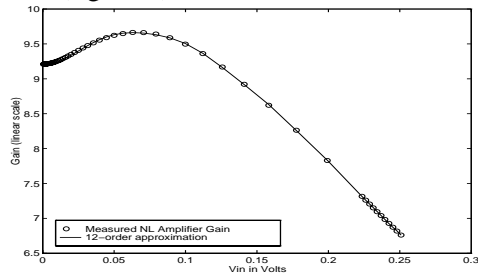


Fig 4: Gain characteristics of the NL amplifier (and its approximation in coincidence)

We have then adapted a second-order pre-distortion system with a neutral initialisation ( $f(|z^2|)=1$ ). After a few thousands iterations we obtain a system which gives the following results expressed in terms of EVM and also ACPR performances.

EVM minimisation. EVM after the pre-distortion system has been tremendously reduced ( $10^{-1}$  at initialisation and  $10^{-5}$  after adaptation). The following figure demonstrates this by means of the constellations before and after adaptation.

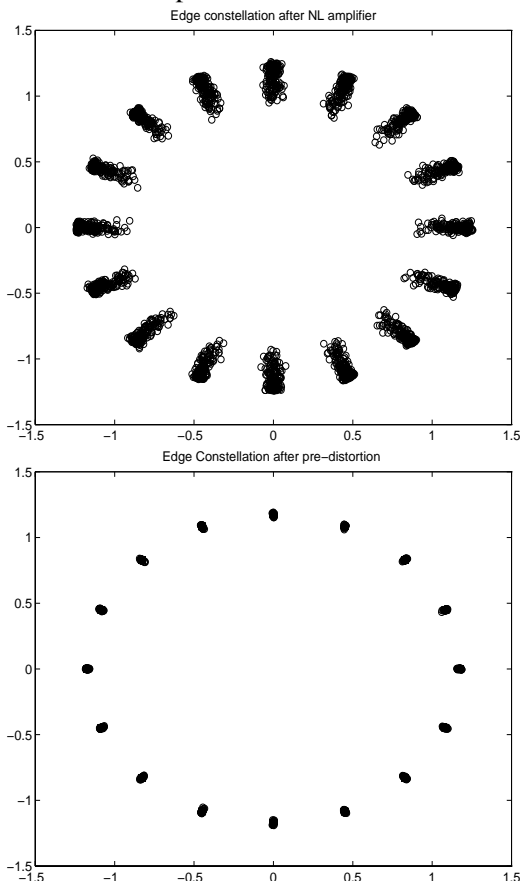


Fig 5 : Edge constellations before and after adaptation of the pre-distortion system.

Spectral consequences. The usual criterion adopted for adaptive pre-distortion is the reduction of ACPR growth due to Non Linear Amplifier. Let us see the power spectral densities (in base band) for the three

signals resulting from amplification of Edge signals :  
 Output of the NL amplifier  
 Output of an ideal (linear) amplifier with constant gain corresponding to the linear range of the amplifier.  
 Output of the adapted pre-distortion followed by the NL amplifier .

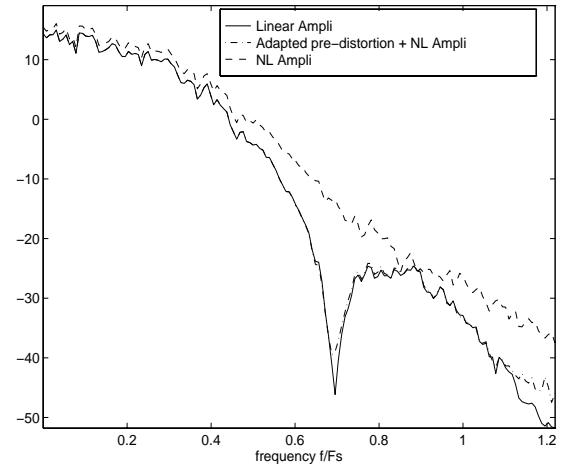


Fig 6: Power Spectral Densities (base-band) before and after pre-distortion adaptation)

We can notice that the ACPR has been drastically reduced whereas it was not our premium objective.

## 6 Conclusions

We have demonstrated an adaptive system of pre-distortion for a non-linear amplifier, based on EVM criterion.

For the moment we have only compensated amplitude distortion (defined by AM-AM curve) and thus we have considered real polynomial functions for the pre-distortion and gain. Moreover we have worked entirely in baseband (using MATLAB tools).

In next step we'll take also AM-PM distortion into account and co-simulate (using ADS tools) the system with the real amplifier.

### References:

- [1] P. B. Kenington, Linearised RF Amplifier and Transmitter Techniques, *Wireless System International*, [http://.www.wsil.com](http://www.wsil.com), 1998.
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