Comparison of Different Adaptation Algorithms for Adaptive Digital Predistortion based on EDGE Standard

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***Abstract* — Radio Frequency (RF) power amplifiers require linearization in order to reduce adjacent channel Inter-Modulation (IM) distortion. Adaptive digital predistortion is one promising linearization technique that can be employed. There are, however, a lot of adaptation algorithms involved with this technique, namely polynomial and look-up table. This paper assesses four methods, which are the combination of Least Square Curve Fitting and Least Mean Square (polynomial) and also Polar, Secant and Linear (look-up table). Their convergence time and mean magnitude of error improvement achieved by using the standard set by EDGE system are compared.**

I. Introduction

Radio Frequency (RF) power amplifiers are widely used in communication systems, particularly in basestation transmitters. If the transmitted signal has non-constant envelope, then Inter-Modulation (IM) distortion will be generated in its nonlinear components. Since these IM powers would interfere with the adjacent channels, it is often desirable to use a highly linear amplifier.

Using a highly linear amplifier, like Class A power amplifier, can indeed reduce the channel nonlinearity, but unfortunately it gives a very low power efficiency. Since most portable equipments nowadays have limited battery capacity, power efficient types of amplifiers offer a great advantage. Thus, in order to employ high power-efficient amplifiers, some kinds of linearization are needed.

There are a lot of linearization techniques, which have been proposed to obtain linear amplification using nonlinear, or high power-efficiency amplifiers. The most commonly used technique to date is the adaptive digital predistortion. The concept behind predistortion is the insertion of a nonlinear module between the input signal and the amplifier. This nonlinear module, called the predistorter has the inverse function of the amplifier, and together with the power amplifier could produce a linear output. The action of this predistorter is achieved by using a DSP (Digital Signal Processor) and it has to adapt to environmental changes like variations in frequency, temperature, power level and component aging. Thus, it is named as adaptive digital predistortion.

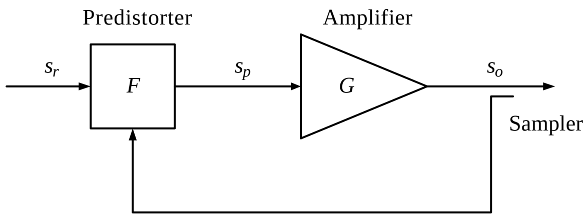

Fig. 1. Simplified model of an adaptive digital predistortion linearizer

EDGE (Enhanced Data rates for GSM Evolution) is a GSM (Global System for Mobile communications) Phase 2+ system, which can provide higher data rates for HSCSD (High Speed Circuit Switched Data) and GPRS (General Packet Radio Service). The use of higher level modulation and higher symbol rate increases the air interface gross rate by a factor of 3, which enables significantly higher data rates. The main application environment for EDGE is an urban environment with quasi stationary or slowly moving mobiles [1].

Generally, there are two forms of adaptation algorithms applied to predistortion, which are polynomial and look-up table. For polynomial, there is a range of methods like Least Square Curve Fitting (LSCF) and Least Mean Square (LMS), whilst look-up table has Polar, Secant and Linear methods. This paper looks at the performance of all these adaptation methods by analyzing their convergence time and mean magnitude of error improvement achieved. The analysis is based on the current EDGE specification [2].

II. Adaptive Digital Predistortion Linearizer

A. General Description

Fig.1 shows a simplified model of an adaptive digital predistortion linearizer, where the predistorter precedes the nonlinear function of the amplifier. To be able to adapt to changes, a fraction of the amplifier output is fed back to the predistorter by using the sampler where the input and output signals are compared and the difference is used to

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adjust the predistortion function. In the sections below, different adaptation algorithms are discussed.

B. Polynomial – LSCF and LMS

Consider a fifth order polynomial function that describes the predistorter, $y(x)$ in terms of the output magnitude of the power amplifier, x with a as the coefficients:

$$y(x) = a_1 x^5 + a_2 x^4 + a_3 x^3 + a_4 x^2 + a_5 x + a_6 \quad (1)$$

By employing the LSCF [3], which is a procedure for obtaining an optimum fit of a functional form to data, the likelihood of obtaining a particular set of N pairs of values (x_i, y_i) in one complete measurement of N points is related to the quantity chi-square, needs to be deduced,

$$c^2(a_1, a_2, a_3, a_4, a_5) = \sum_{i=1}^{N} \left(\frac{y_i - y(x)}{\Delta y_i} \right)^2 \quad (2)$$

where the known standard deviation of the measurement of the i th data points is assumed to be approximately the same as the experimental errors associated with each measurement, i.e. $s_i = \Delta y_i$. Minimizing this expression will yield the optimum set of parameters for this particular choice of functional form. Thus, the five equations for the minimum of c^2 are:

$$\frac{\partial c^2}{\partial a_1} = \frac{\partial c^2}{\partial a_2} = \frac{\partial c^2}{\partial a_3} = \frac{\partial c^2}{\partial a_4} = \frac{\partial c^2}{\partial a_5} = 0 \quad (3)$$

In matrix notation, these conditions yield

$$\begin{pmatrix} \overline{x^{10}} & \overline{x^9} & \overline{x^8} & \overline{x^7} & \overline{x^6} & \overline{x^5} \\ \overline{x^9} & \overline{x^8} & \overline{x^7} & \overline{x^6} & \overline{x^5} & \overline{x^4} \\ \overline{x^8} & \overline{x^7} & \overline{x^6} & \overline{x^5} & \overline{x^4} & \overline{x^3} \\ \overline{x^7} & \overline{x^6} & \overline{x^5} & \overline{x^4} & \overline{x^3} & \overline{x^2} \\ \overline{x^6} & \overline{x^5} & \overline{x^4} & \overline{x^3} & \overline{x^2} & \overline{x} \\ \overline{x^5} & \overline{x^4} & \overline{x^3} & \overline{x^2} & \overline{x} & 1 \end{pmatrix} \cdot \begin{pmatrix} a_1 \\ a_2 \\ a_3 \\ a_4 \\ a_5 \\ a_6 \end{pmatrix} = \begin{pmatrix} \overline{x^5 y} \\ \overline{x^4 y} \\ \overline{x^3 y} \\ \overline{x^2 y} \\ \overline{xy} \\ \overline{y} \end{pmatrix} \quad (4)$$

where the averages are calculated as

$$\overline{x} = \frac{1}{N} \sum x_i, \overline{y} = \frac{1}{N} \sum y_i, \overline{x^n y} = \frac{1}{N} \sum x_i^{\,n} y_i \quad (5)$$

The solution to this matrix equation will return the values of the coefficients, a_k, of the best-fit quadratic to the data. Since for adaptive digital predistortion, the data points are not obtained all at once, but more accurately, they are computed at each sample time, so the calculation of the averages shown in (5) has to be changed to use an algorithm based on the LMS. A matrix element called w is defined as follows, with k denoting the sampling instant.

$$w(k+1) = w(k) - b \cdot (w(k) - v) \quad (6)$$

where v represents x, y and xy in (5), b controls the stability and rate of convergence of the algorithm and it can be altered to optimize the system's performance.

C. Look-Up Table – Polar

Consider the system in Fig. 1. If the amplifier is nonlinear, the amplitude predistortion gain [4] is given by

$$|F_i(k+1)| = |F_i(k)| - a_R \cdot (|s_o| - |s_r|) / |s_o| \cdot |F_i(k)| \quad (7)$$

where $F_i(k)$ denotes the k th iteration of table entry i, with $|s_o|$ and $|s_r|$ being the output and input amplitudes respectively. a_R is the positive adaptation factor, which has bounds as stated below

$$a_R < 2 \cdot \left(\frac{|G|}{\partial |s_o| / \partial |s_p|} \right), \text{with } G = \frac{s_o}{s_p} \quad (8)$$

The phase adaptation can be applied directly, that is

$$\angle F_i(k+1) = \angle F_i(k) - a_q \cdot (\angle s_o - \angle s_r) \quad (9)$$

where a_q is the adaptation factor which must be less than 2 for stability and equal to 1 for optimum convergence [4].

D. Look-Up Table – Secant

This method [5] has the iteration as shown below:

$$F_i(k+1) = F_i(k) - e_g(F_i(k)) \cdot Y \quad (10)$$

where

$$Y = \frac{F_i(k) - F_i(k-1)}{e_g(F_i(k)) - e_g(F_i(k-1))} \quad (11)$$

In order to solve the instability and adaptation jitter problems, the error calculation was altered [5] to:

$$e_g(F) = s_o \cdot \frac{|s_r|}{s_r} - |s_r| \quad (12)$$

E. Look-Up Table – Linear

This method [4] is similar to the polar adaptation algorithm, the only difference comes from the fact that the operation now is in complex gain.

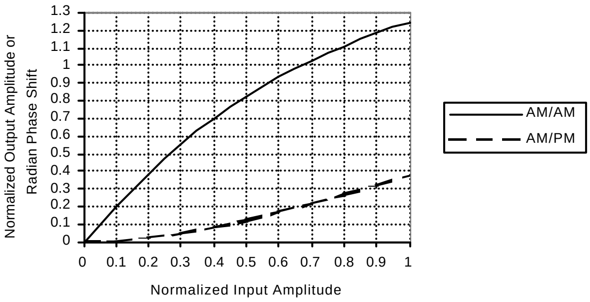

Fig. 2. Class AB power amplifier

Again, consider the system in Fig. 1, and if the amplifier is nonlinear, the correction on every table entry is,

$$F_i(k+1) = F_i(k) - a_{opt} \cdot ((s_o - s_r)/s_o) \cdot F_i(k) \quad (13)$$

where a_{opt} is the optimum complex adaptation factor [4] which is calculated as

$$a_{opt} = \frac{G}{\partial(s_p G)/\partial s_p}, \text{ with } G = \frac{s_o}{s_p} \quad (14)$$

III. SOFTWARE SIMULATION

A. General Description

The simulation is modeled using SIMULINK while numerical calculation is programmed using MATLAB. It should be noted that uniform random numbers are chosen as the system's input. Also, up- and down-conversions are not included: all operations are performed at baseband.

A class AB power amplifier model is used throughout the simulation owing to its high power efficiency as well as low output signal distortion, which is very suitable for portable equipment with limited battery capacity. The AM/AM (amplitude dependent gain) and AM/PM (amplitude dependent phase shift) plots of this amplifier are shown in Fig. 2.

B. Calculation of Error Vector Magnitude

The relative RMS (Root-Mean-Square) vector error [2] is defined as

$$RMS\ EVM = \sqrt{\frac{\sum_{k \in K} |E(k)|^2}{\sum_{k \in K} |S(k)|^2}} * 100\% \quad (15)$$

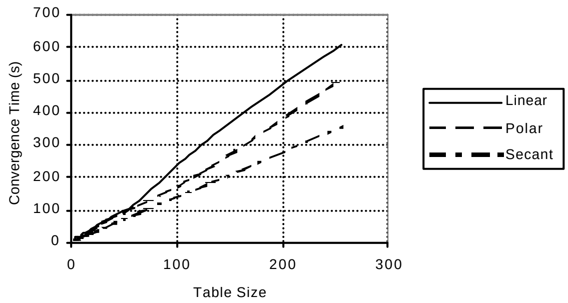

Fig. 3. Convergence time as a function of table size for look-up table adaptation algorithms

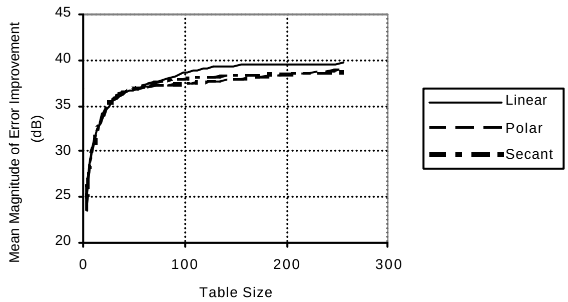

Fig. 4. Mean magnitude of error improvement as a function of table size for look-up table adaptation algorithms

where k is the sample time, $S(k)$ is the amplifier's output and $E(k)$ is the difference between the amplifier's output and the predistorter's input.

According to [2], the measured RMS EVM over the useful part of any burst shall not exceed 7% under normal conditions (excluding the effect of any passive combining equipment) for Base Transceiver Stations (BTS).

C. Convergence Time Measurement

In order to measure the convergence time of all the adaptation algorithms mentioned in the previous section, the simulation is conducted by running the simulation until the RMS EVM reaches less than or equal to 7%. Since the RMS EVM will be decreasing from the start of the simulation owing to the adaptation of the system, the point where the RMS EVM reaches 7% is taken as the reference point for the convergence time for a particular adaptation algorithm employed.

Different table sizes of 2 to 256 have been tested to see the effect on the convergence time and all three look-up table methods have been simulated. As can be seen from Fig. 3, convergence time is linearly proportional to the increase in table size (except for linear and polar methods, where both have sudden increment in gradient after certain table sizes). The secant method has the shortest convergence time for every table size, with the polar method second, while the linear method has the worst performance of the three.

TABLE I

COMPARISONS OF THE ADAPTATION ALGORITHMS

Analysis	Adaptation Algorithms			
	Polynomial (LSCF and LMS)	Look-Up Table (Table Size 32)		
		Linear	Polar	Secant
Convergence Time	19.51s	68.24s	66.43s	47.27s
Mean Magnitude of Error Improvement	47.669dB	35.877dB	36.034dB	36.017dB

D. Mean Magnitude of Error Improvement Measurement

In order to study the effect of the number of table entries on the accuracy of the predistortion function, the mean magnitude of error improvement measurement is calculated. This is deduced by subtracting the mean magnitude error of the non-adaptive system (without predistortion) from the adaptive system (with predistortion) when the RMS EVM of the adaptive system reaches 7%, which is simply the error improvement.

From Fig. 4, the mean magnitude of error improvement increases with table size but with a decreasing gradient (except for linear method, where it has a sudden increment in gradient after a certain table size). When the table size is between 2 and 32, all methods perform equally well, but when it is more than 32, the linear method dominates, followed by the secant and polar methods.

One can notice that when the table size is increased from 32 to 256, the error improvement is just around 4dB, whilst the convergence time can increase up to 600s (refer to Fig. 3 and Fig. 4). This shows that for larger table size, the predistorter does give a slightly more accurate function, but at the expense of longer convergence time.

E. Comparison of Look-Up Table and Polynomial Adaptation Algorithms

The polynomial (LSCF and LMS) adaptation algorithm has also been tested with these convergence time and error improvement computations.

By looking at Table I, we can see that polynomial adaptation algorithm outperforms the look-up table adaptation algorithms. The convergence time of the polynomial is almost three times faster than that of the look-up table, whilst the mean magnitude of error improvement is about 11.7dB more in average. Among look-up table method, the secant method has the best performance overall owing to its fast convergence time, followed by polar and linear methods. Note that table size of 32 has been chosen for the look-up table adaptation algorithms.

IV. CONCLUSION

In this study, the calculation of RMS EVM has been implemented, where the percentage ratio is set according to the EDGE specification, to test the convergence time and mean magnitude of error improvement for polynomial and look-up table adaptation algorithms. A table size of 32 has been chosen owing to its high mean magnitude of error improvement with acceptably low convergence time. The secant method shows the best performance overall among all the look-up table method in convergence time and error improvements, with polar method second and linear method at last. However, these look-up table methods still cannot match with the polynomial (LSCF and LMS) adaptation algorithm, where the latter has far better convergence time and error improvement than the former.

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