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電力増幅器用適応ディジタルプリディストーションリニアライザ

ダシティ[†,††]　モハマディ[†]　高田　潤一[††]

† アミルカビル工業大学
イラン テヘラン市
†† 東京工業大学 大学院理工学研究科
〒 152-8550 東京都目黒区大岡山 2-12-1
E-mail: †{dashti,takada}@ap.ide.titech.ac.jp

あらまし　多値 QAM 変調は周波数利用効率が高い一方，大きな包絡線変動のため，電力増幅器 (HPA) が飽和状態に近い非線形領域で動作すると帯域外成分が増大するという問題がある．本研究は，テーブル参照 (LUT) 方式に基づいた適応プリディストーション (PD) アルゴリズムを用いて PA の線形化を図った．本アルゴリズムを AB 級 HPA の実測結果に適用してその効果を評価した．隣接チャネル漏洩電力比 (ACPR) を用いて PD の性能評価を行った結果，16-QAM 信号を入力した場合に，PD の有無で出力スペクトルを比較した結果，PD 使用時に ACPR が約 20 dB 改善されることを確認した．

キーワード　プリディストーション，電力増幅器，リニアライザ，ACPR

Adaptive Digital Predistortion Linearizer for High Power Amplifiers

M. DASHTI[†,††], A. MOHAMMADI[†], and Jun-ichi TAKADA[††]

† Amirkabir University of Technology
Tehran, 15914, Iran
†† Graduate School of Engineering, Tokyo Institute of Technology
2-12-1 O-okayama, Meguro-ku, Tokyo, 152-8550 Japan
E-mail: †{dashti,takada}@ap.ide.titech.ac.jp

Abstract　M-QAM modulation has been considered to achieve high bandwidth efficiency for wireless communications. However, due to its envelope fluctuation, it exhibits large spectral regrowth when power amplifier (PA) operates in nonlinear region close to saturation. in this study, an adaptive predistorter (PD) algorithm, based on look-up table (LUT), is introduced to linearize PA. The algorithm is implemented for a measured Class AB HPA. Adjacent Channel Power Ratio (ACPR) is used as a criterion to evaluate the PD performance. Output spectrum with and without PD are compared for 16-QAM input signal. A nearly 20 dB improvement in ACPR is obtained by using PD.

Key words　Predistortion, Power amplifier, Linearization, ACPR

1. Introduction

The final amplifying stage of a radio transmitter usually consumes a considerable amount of power. For mobile communication it is necessary for this stage to maximize the efficiency of the power amplifier; which means that the PA must work near saturation. Constant envelope modulation such as GMSK allow PA to operate in the nonlinear region near saturation, for power efficiency, yet they do not generate inter modulation products in nearby channels. Unfortunately these modulation schemes are less spectrally efficient than linear modulation such as QAM which produce variation in both phase and envelope of the signal. Considering envelope fluctuation in the transmitted signals, when they pass through a nonlinear PA, they cause the spectrum of the signal to expand into adjacent channels producing interference for other users. Considering stringent restrictions on out of band emissions in mobile communication, designers are faced with two alternatives; back off the PA to a linear operating region even more inefficient; or linearize the amplifier.

A number of linearization techniques have been reported in recent years. Many of them suffer from limitations in bandwidth, precision or stability. One method of linearization that can compensate for PA nonlinearities in such an environment is to predistort the drive signal [1]~[3], [6], [8], [9]. The concept is based on inserting a nonlinear function with the inverse function of PA between the input signal and the amplifier to produce a linear output. This can be achieved in base band by using look up table technique. If accurate predistortion is required it is necessary to adjust the predistort signal so that it can track changes in amplifier characteristics.

In [1], [3] two gain based fast adaptive predistorters were introduced by exploiting a low memory look up table. In [1], a Power-based LUT is applied to produce the output I/Q samples, this method is well suited for DSP implementation. However, the In-phase and Quadrature errors are dependent and the adaptation is formulated as a root finding problem and the proposed secant method to update the table, reduces the convergence rate .In [3], Polar representation model is considered for the input samples and an amplitude based LUT is used. This method shows better performance and very fast convergence at the expense of implementing rectangular to polar and polar to rectangular conversions. In [4] an adaptive predistortion method using a LUT technique presented, called slope dependent method, and faster convergence was reported in comparison with the direct linear LUT used by Caver [1]. In [10], a Polynomial model used for the Predistorter in order to compensate for the AM/AM and AM/PM characteristics of the amplifier and a simple and fast converging method obtained by using LMS algorithm to estimate the polynomial coefficients. An I/Q represented polynomial method based on adjacent channel emission measurement was introduced in [6]. Another adaptive polynomial I and Q predistorter was presented in [11], which shows a good convergence rate and a high reduction in spectrum spreading. In [11] a combination of analog predistorter and postdistorter was used to improve the convergence rate.

This paper describes an adaptive digital base band predistortion technique based on LUT for linearizing a typical class AB high PA. A polar representation form is considered for the input samples and amplitude based LUT is used. An adaptation algorithm based on slope of the PA characteristics is introduced to adapt PD with chang of the PA characteristics. Based on the optimum obtained values of table size and convergence factor, a relatively high level adaptation is achieved. Simulation results show significant reduction in adjacent channel power ratio (ACPR) and error vector magnitude (EVM). The proposed predistortion system is overviewed with emphasis on the operation of LUT in section 2 . The results through simulation of applying the PD system to a 16QAM signal are presented in section 3. The algorithm is evaluated concerning its convergence rate and ACPR. The implementation simplicity aspect of this method is then discussed. Brief conclusions follow in section 4.

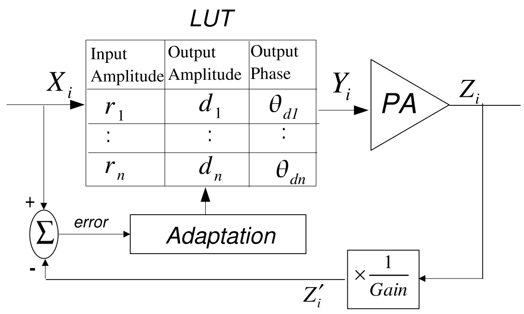

Figure 1 baseband model for the adaptive predistorter

2. Addaptive digital baseband predistortion

The block diagram of an adaptive base band predistorter is shown in Fig. 1.

The system input can handle any type of modulation. For each input value, X_i, the algorithm search trough the table to find the best matched value,Y_i . Hence i-th input sample value,$X_i = r_i e^{j\theta_i}$, is converted to predistorted value ,$Y_i = d_i e^{j(\theta_i + \theta_{di})}$ in Fig.1.1. Note the peak value of $|X_i|$ must be scaled equal to the address size of the table.

The algorithm should be able to identify the amplifier characteristics and update the table to adapt with the variation of PA characteristics. In this work, table updating based on minimizing the error, defined as the difference between input, X_i , and normalized output ,Z'_i , is done.

Input samples in PD can be presented in two different format; rectangular (IQ format) or polar representation formats. Cavers introduced an adaptive predistorter by exploiting a low memory LUT in rectangular form [1]. This method is well suited for DSP implementation. However, in-phase (I) and Quadrature (Q) errors are dependent and the adaptation is formulated as a root finding problem. The proposed secant method to update the table, reduces the convergence rate and therefore degrades the overall performance.

In this work, a polar represented table is used to make the complex function for the predistorter. Since the phase and amplitude error are independent, this method seems to provide faster convergence rate at the expense of implementing rectangular to polar and polar to rectangular conversions.

2.1 Table operation

The operation of the table in both access and adaptation mode is described below.

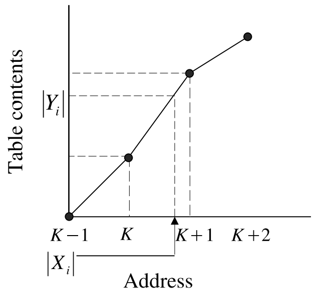

Figure 2 Table access

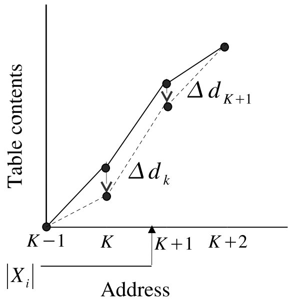

Figure 3 tabel adaptation

Table access

The operation of the amplitude table in access mode is shown in Fig. 2. The i^{th} desired input value, $|X_i|$, should be converted to predistorted input to HPA, Y_i. Since the table is not large enough to save all the possible input amplitudes value, $|X_i|$ is unlikely to fall directly on a table entry, and so the predistorted output value is determined by a linear interpolation between the nearest address values e.g. if $r_k \leqq |X_i| < r_{k+1}$ then the table output amplitude and phase will be found using linear interpolation accroding to the following line:

$$|Y_i| = d_k + \frac{d_{k-1} - d_k}{r_{k+1} - r_k}(|X_i| - r_k) \tag{1}$$

$$\angle Y_i = \theta_{dk} + \frac{\theta_{dk+1} - \theta_{dk}}{r_{k+1} - r_k}(|X_i| - r_k) + \angle X_i \tag{2}$$

where d_k is the contents of the k-th location in the table and r_k is the largest whole number less than $|X_i|$. The predistorted value,Y_i, then passes through the nonlinear power amplifier to generate the amplified signal , Z_i.

Table adaptation

The output amplified signal, Z_i, passes through the scalar block with the inverse desired gain of PA , to generate envelope normalized signal ,Z'_i (Fig. 1). The feedback value,Z'_i , should be equaled to the desired input,X_i .When this is not so due to the variation of PA characteristics, adaptation of the table values should occur. Feedback value, Z'_i, is contrasted with the appropriate input signal , X_i, to provide the error to feed the algorithm and update the table. The amplitude correction value is

$$\Delta d = S_a(|X_i| - |Z'_i|) \tag{3}$$

Where $(|X_i| - |Z'_i|)$ is the error, and the constant,S_a , determines stability, convergence rate, and the sensivity of the system to external errors. In this work, the correction, Δd, is apportioned to the table entries whose addresses are on either side of the input value,$|X_i|$ as shown in Fig. 4, where $r_k \leqq |X_i| < r_{k+1}$. The addresses closest to the input value gets the largest share of error .The updating value is weighted as shown by the following equations.

$$\Delta d_k = (\frac{r_{k+1} - |X_i|}{r_{k+1} - r_k})\Delta d \tag{4}$$

$$\Delta d_{k+1} = (\frac{|X_i| - r_k}{r_{k+1} - r_k})\Delta d \tag{5}$$

Thus the k-th and $k+1$-th entries of the table are updated as:

$$d_k \leftarrow d_k + \Delta d_k \tag{6}$$

$$d_{k+1} \leftarrow d_{k+1} + \Delta d_{k+1} \tag{7}$$

In this work, we update only the two nearest points by each samples to avoid increasing the table updating computations. However increasing these updating points to four make the algorithm more stable and faster to convergence .The same steps are done for phase table updating .The phase correction value is

$$\Delta\theta = S_p(\angle X_i - \angle Z'_i) \tag{8}$$

where $(\angle X_i - \angle Z'_i)$ is the error in i-th sample and the parameter S_p is to control the convergence rate and stability. By weighting the updating value, same as amplitude updating, k-th and $k+1$-th entries of the table are updated as:

$$\theta_k \leftarrow \theta_k + (1 - (|X| - r_k))\Delta\theta \tag{9}$$

$$\theta_{k+1} \leftarrow \theta_{k+1} + (|X| - r_k)\Delta\theta \tag{10}$$

For the small enough value of $\Delta\theta$, feedback amplitude is changed from $|Z'|$ to $|Z'| + \Delta|Z'|$, where the error reduction is $\Delta|Z'| = \frac{\partial|Z'|}{\partial d}\Delta d$ and $\frac{\partial|Z'|}{\partial d}$ is the differential gain of amplifier. The convergence rate is therefore dependent on the amplifier characteristics [3]. The stability depends on the dynamic slope of the AM/AM characteristics of the power amplifier and the desired gain. Stability is guaranteed if

$$\Delta|Z'| \leqq 2(|X| - |Z'|) \tag{11}$$

which gives

$$\begin{aligned} S_a &\leqq \frac{2}{\max(\frac{\partial|Z'|}{\partial d})} \\ &\leqq \frac{2}{\text{maximum differential gain of amplifier}} \end{aligned} \tag{12}$$

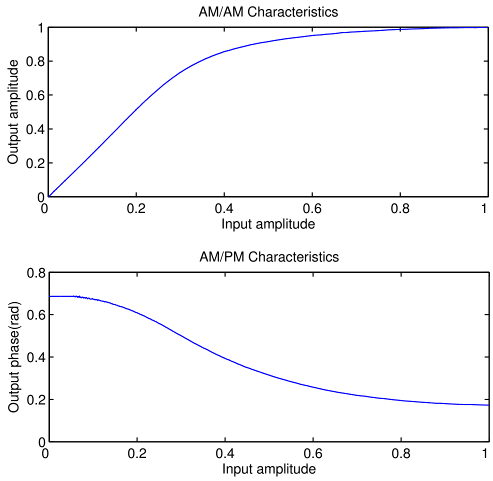

Figure 4 AM/AM and AM/PM characteristics of PA

The optimum speed of convergence for a given differential gain occurs at half of this value. smaller values reduce the convergence rate but damp out external influences such as noise and misalignments generated in the feedback path. This method of convergence is slow where there is a low differential gain as occurs in the compression and low signal regions of the amplifier characteristics. In fact the slope of the power amplifier is not constant; and at the start of the adaptation we could be operating around one region while at the end of adaptation we might be operating around another. Therefore S_a has to be variable [4]. In this work for simplicity it is considered to be constant and is taken as the maximum slope of the curve.

The same steps can be described for the phase convergence. The phase adaptation constant, S_p, must be less than 2 for stability and equal to 1 for quickest convergence [3]. It is worth to mention that, the amplitude stability is the necessary condition for the phase convergence.

3. Simulation results

A typical class AB HPA measured model is used for nonlinear PA. A block of input and output 16 QAM samples is used to measure the characteristics of the PA. The smooth and noiseless version of this measurement is used for the simulation. The nonlinear PA suffers from two source of nonlinearity; amplitude to amplitude distortion(AM/AM) and amplitude to phase distortion(AM/PM). Fig. 4 shows the normalized measured AM/AM and AM/PM characteristics of PA used in this simulation. It is assumed that the PA nonlinearities are memory less. [7] Linear interpolation is used to find the PA characteristics curves and since there are not enough data for very low amplitudes, linear extrapolation is used to span data for this out of range values.

All the simulations are performed in base band. The tested signal is 16QAM input signal with over sampling rate of 16 sample per QAM symbol.

At the beginning the amplitude table is assumed to be transparent and all output phase shift are set to zero (i.e. Y(t)=X(t)).

Uniform spacing of the LUT entries was employed in this simulation. Making Predistortion tables equispaced seems to be an excellent choice from an engineering point of view: it is simple, its performance is close to that of defined by optimum spacing method , and it does not dependent on amplifier modulation format [5].

The optimum values of table size and convergence factor were obtained by simulation. The table size is optimized considering convergence rate and error. Various table sizes are evaluated in the simulation from 16 to 128. It was seen that small table size causes faster convergence rate but increases the steady state error and large table size decrease convergence rate. The best value which provided both sufficiently small error and fast convergence was found to be 64.

In order to evaluate the convergence rate, relative mean squared error (mse) criterion between the desired and feedback signals are widly used in the literature. Here, error vector magnitude (EVM) is considered as a criterion to evaluate the proposed predistortion performance, convergence rate and constellation quality.EVM is defined as:

$$EVM = \sqrt{\frac{\frac{1}{N}\sum_{j=0}^{N-1}(\Delta I_j^2 + \Delta Q_j^2)}{\frac{1}{N}\sum_{j=0}^{N-1}(I_j^2 + Q_j^2)}} \tag{13}$$

where ΔI and ΔQ are the magnitude errors in the received symbols.N is the number of samples.

Various values of S_a from 0.1 to 1, for a table size of 64 were investigated using the proposed adaptation algorithm. By changing S_a from 0.1 to 0.8, EVM decrease from 0.0045 to 0.002. Convergence factor larger than 0.8 increase EVM. Therefore, $S_a = 0.8$, was found by simulations to be a good choice. Figure 5 shows the effect of convergence factor, S_a, on Error vector magnitude after 1000 updates for 30dB input signal. From this figure, it can be clearly shown that $S_a = 0.8$ is a good choice for convergence factor.

Significant reduction in Error Vector Magnitude was obtained using proposed PD.EVM was reduced from 0.0063 to 0.00095 respectively after 100 to 5000 updates. Fig. 6 shows the convergence of table (EVM versus number of updates for a 64 entry LUT). From the figure EVM was reduced by more than 40dB after the convergence i.e. about 5000 samples with a selected convergence factor of $S_a = 0.8$. A relative mse of -40dB is reported to be adequate to guaran-

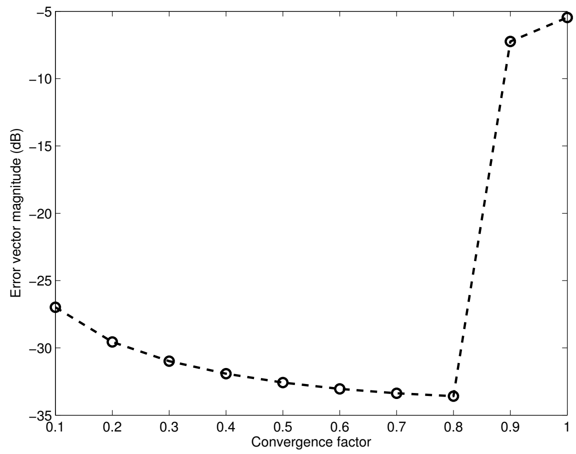

Figure 5 The effect of convergence factor on EVM

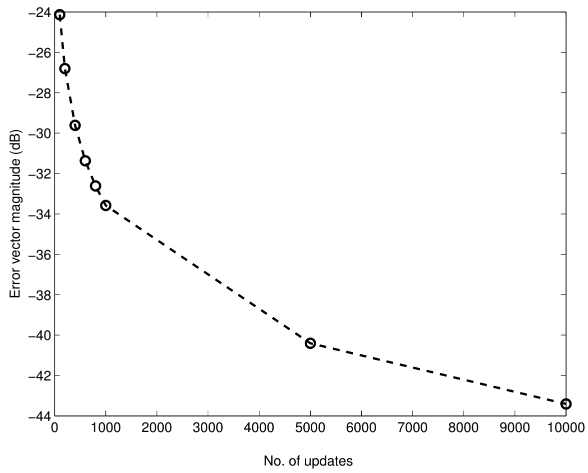

Figure 6 EVM versus number of updates for a 64 entry table

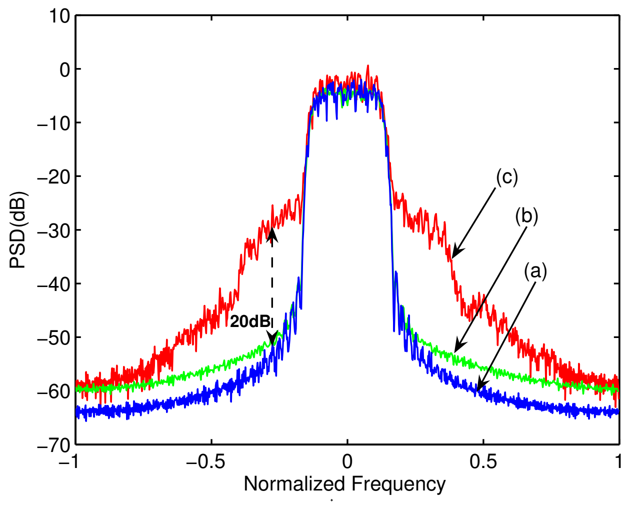

Figure 7 Effectiveness of Predistortion in suppressing spectral regrowth (a)Original input (b)Output spectrum with Predistortion, (c)Output spectrum without Predistortion.

tee an Adjacent Channel Interference (ACI) level of -60dB for most modulation [3]. Considering the ACI requirement for a particular application, the required number of updates would be in the range of 400 to 5000.

The proposed adaptation method is based on the slope of PA characteristics. It is reported in [4] that slope dependent adaptation method provide faster convergence rate and less residual error after convergence in comparison with direct convergence method presented in [1].

A common criterion to measure the linearity of the PA is Adjacent Channel Power Ratio (ACPR), defined as the ratio between the power of the adjacent channels and the power of the main channel. Figure.7 illustrates the output spectrum with and without predistorter in comparison to the input signal. As figure shows, the non-linear amplifier causes spectral regrowth as well as in-band distortion. Using predistorter, it is seen that about 20 dB improvement in adjacent channel power ratio is obtained.

Assessment of a predistorter must takes into account both its performance and complexity. At standpoint of performance, as presented above, the proposed method provide significant reduction in ACPR and EVM.

Another advantage of using this method is its simplicity to implement since the number of mathematical operations involved in each iteration is not large. As presented in Eqs.(3)-(10), there is no division in adaptation equation. Therefore it is simpler than direct convergence method described by Cavers which has division in its update equation.

4. Conclusion

This paper presented an adaptive digital base band predistortion method using a polar represented LUT for linearizing a nonlinear power amplifier. An adaptation method based on the slope of the PA characteristics is then introduced for updating the LUT with PA characteristics changes. Since the proposed adaptation method does not contain any division or large number of mathematical operation in each update iteration, It is simple to implement. The system is tested by the simulation class AB HPA with 16QAM input. Algorithm performance is evaluated concerning its convergence rate by EVM. Simulation results based on the ACPR and EVM criteria to evaluate PD performance were presented. The results show significant reduction in adjacent channel power ratio and EVM.

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