# ADAPTIVE DIGITAL PREDISTORTER FOR POWER AMPLIFIERS WITH REAL TIME MODELING OF MEMORYLESS COMPLEX GAINS

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## ABSTRACT

When the resulted signal from linear modulation methods, like M-ary QAM, are passed through a nonlinear power amplifier, their fluctuating envelope causes distortion and spectral spreading. In order to avoid these effects, maintaining both power and spectral efficiency, lead to use of linearization techniques. This paper presents a digital predistorter with real time modeling of AM-AM and AM-PM characteristics of a power amplifier (PA). The input and output lowpass equivalent complex envelopes of the amplifier are sampled, scaled and updated into a lookup table to provide the predistorted signal. An improvement of 45 dB of out-of-band power is obtained in simulating with Signal Processing WorkSystem (SPW). Then, convergence time is eliminated. The proposed technique is robust and efficient since no iterative procedure is needed, hence the convergence time is eliminated

## I INTRODUCTION

With the increasing demands on the RF and microwaves spectrum, caused by the proliferation of wireless communications and satellite network, more spectrally efficient modulation techniques will have to be used. Linear modulation method, like M-ary QAM, meets this requirement, but their performance is strongly dependent on the linearity of the transmission system. In addition, to maximize the high power added efficiency and the power output, the power amplifier is often operated near saturation where the input/output relationship became nonlinear, if linear modulation with fluctuating envelope is used, distortion and spectral spreading into adjacent channels that causes interference for other users will be occurred. In order to reduce these undesired effects and meeting power and spectral efficiency as required, linearization techniques have been introduced.

A variety of linearization methods have been reported and the three main types are: 1)Feed-forward [1], it includes an open loop configuration and, appropriate for wideband, it can handle multicarrier signal but can not easily be controlled against the effects of drift, moreover, their poor efficiency make it suitable in basestation only. A good analysis of adaptation behavior has been presented in [2]. 2)Cartesian-Feedback [3], it presents an excellent reduction of out-of-band emissions and it is relatively easy to implement, however stability requirement limits its bandwidth because of critical dependence on the loop delay. 3) Predistortion [4], having an open loop configuration, it uses a nonlinear element that, preceding the device to be compensated, its gain expansion characteristic cancel the gain compression of the amplifier that yields the linear overall transfer function of both, the predistorter and the amplifier. Like feed-forward, it is very sensitive to drifts.

Recently, the technology progress of digital signal processors has been one of the motive of the imminent course toward digital modulation techniques. Various applications such as realization of digital filters, generation of accurate gain and phase matching in two quadrature modulating signal and real-time compensation of amplifier non-linearities have permitted the use of these processors in several methods of linearization, which are called "Digital Linearization Techniques".

In order to avoid the effect of temperature variation, device power supply precision and drifts produced by switching between channels, adaptability is needed. In this case, Adaptive Digital Predistorter is the most promise technique that can be applied to narrow band Personal Communication Service (PCS) using a digital signal processor (DSP). The first successful work were presented by Nagata[5] using a two-dimensional look-up table technique with adaptive digital feedback at baseband and pulse shaping filter prior to predistortion. This technique has the advantage that any order of nonlinearity and any modulation format can be performed. Followed later by Caver[6] and Faulkner[7], several drawbacks were improved using one-dimensional table, it has made possible that less memory is needed and reduces the convergence time. These previous techniques used iterative algorithm.

An interesting idea was proposed by Wilkinson [8] using two look-up tables (amplitude and phase) with 100 entries in each table covering the range of input levels. Linear interpolation is used to interpolate between entries which reduce the size of memory required. The later technique does not consider any adaptability dedicated to drift correction.

We describe in this paper an Adaptive Digital Predistorter with Real-Time Modeling of memoryless





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complex gains that can supply correction without using iterative algorithm and without any restriction as the loop delay. Moreover, any order of nonlinearity and any modulation format can be suppled by the linearizer. This technique, which uses the concepts of lowpass equivalent signals and systems, is implemented with a digital signal processor where the Real-Time Modeling of AM-AM and AM-PM characteristics is performed to provide, using a one-dimensional lookup table technique, the predistorted signal.

## II THE NEW LINEARIZATION TECHNIQUE

Fig.1 shows a block diagram of the adaptive predistortion linearizer where in addition to the digital domain, a dual DA/AD converters, quadrature modulator and demodulator, microwave coupler and microwave power amplifier form the analog domain to complete the global system. The spectrally efficient 16QAM modulation method is used as signal source, which is passed through a pulse shaping filter to ensure free inter-symbol-interference (ISI). The input and output lowpass equivalent complex envelopes of the amplifier are sampled, scaled and updated into a lookup table to provide the predistorted signal. Note that cascade form was chosen to generate amplitude and phase predistortion. Feedback is used only to update the look-up table. The feedback loop is opened until new significant drifts are occurred and new data has to be entered in the look-up table. In this case, mean error between the desired and the distorted feedback signals criterion is used to performs adaptability. In order to compensate feedback path's delay, the samples of the input signal are delayed through a delay block.

For modeling propose, the amplifier is considered a memoryless nonlinearity [9] and its complex gain as function of the input amplitude only, hence the possibility to use one dimensional table. The amplitude transfer characteristic AM-AM and AM-PM conversion factor of the PA used, are shown in fig. 2. Let the amplifier input signal be:

$$S(t) = V_m(t)\cos[w_c t + \theta(t)] \quad (1)$$

then the output signal may be expressed by:

$$Z(t) = G[V_m(t)]\cos\{w_c t + \theta(t) + \Phi[V_m(t)]\} \quad (2)$$

where

$G[V_m(t)]$ is the AM-AM characteristic
$\Phi[V_m(t)]$ is the AM-PM characteristic

If we consider the predistorter output, which is the amplifier input signal, as follows:

$$V_d(t)\cos\{w_c t + \theta(t) + \alpha[V_d(t)]\} \quad (3)$$

Then the output signal of the PA will be

$$G[V_d(t)]\cos\{w_c t + \theta(t) + \alpha[V_d(t)] + \Phi[V_d(t)]\} \quad (4)$$

From (1) and (4), we can see that the conditions that must be satisfied to perfectly correction of the AM and PM distortion are :

$$G[V_d(t)] = KV_m(t) \quad (5)$$

$$\Phi[V_d(t)] + \alpha[V_d(t)] = 0 \quad (6)$$

where K is the constant gain in the linear region and $V_m(t)$ is the amplitude of the input signal to be amplified.

## III SIMULATION METHOD

Complex Envelope Method [10], which uses the concepts of lowpass equivalent signals and systems, has been used to analyze and simulate the system. We can write any x(t) modulated carrier signal as:

$$x(t) = \mathrm{Re}[V(t)\, e^{j[w_o t + \phi(t)]}] = \mathrm{Re}[V(t)\, e^{j\phi(t)}\, e^{jw_o t}] \quad (7)$$

where the complex envelope of bandpass signal or lowpass equivalent complex signal is given by:

$$V(t)\, e^{j\phi(t)} \quad (8)$$

- $V(t)$ is the amplitude modulation
- $\phi(t)$ is the phase modulation of the signal

In this manner, (8) can be traited as an equivalent lowpass signal if the bandwidth B is much lower than the carrier frequency $w_c$, that is, $B \ll w_c$.

Polar representation was chosen to configure the look-up tables and these can be accessed in cascade form where the first and the second table generate the predistorted amplitude and phase respectively. These tables implement a mapping from the input to the output, according to the number of sampled pairs, using linear or spline interpolation. If spline interpolation is used, only a small number of sampled pairs are needed. In order to estimate the delay in the feedback loop, correlation between input amplifier and feedback signal is performed and the delay is compensated by the same amount in the input signal. The oversampled rate was 16 samples/symbol and the pulse shaping filter was a squarer root raised cosine with 0.35 roll off. Moreover, gain unbalance, phase unbalance and carrier feed-through of the quadrature modulator and demodulator are assumed to be zero. Code Generation System [10] from SPW was used to produce automatically optimized C code of the algorithms. After compilation, the code was ran in a supported DSP Texas Instruments (TMS320C40).

## IV PRELIMINARY RESULTS

As described previously, the global system has been validated on SPW environment and several important analysis results are presented here. Fig.3 shows 38 samples of the input and output lowpass equivalent complex envelopes of the amplifier that are used for modeling. It shows also the amplitude of memoryless complex gain and phase predistortion. Fig.4 show the power spectral density with and without linearization, an improvement of 45 dB of out-of-band power is obtained. Since interpolation is used, convergence time is eliminated.

## V CONCLUSION

We proposed a new method dedicated to Adaptive Digital Predistorter with real time modeling of AM-AM and AM-PM characteristics. Power efficiency, spectral efficiency and signal quality are some important goals in wireless communication and satellite network. Reaching all these goals, when linear modulation is employed through nonlinear system, need some compensation for nonlinearity. Our proposed linearization technique considers adaptability through drift correction and eliminates an iterative procedure. Also, any order of nonlinearity and any modulation format can be performed. The preliminary results motivated us to plan the realization of a portable system.

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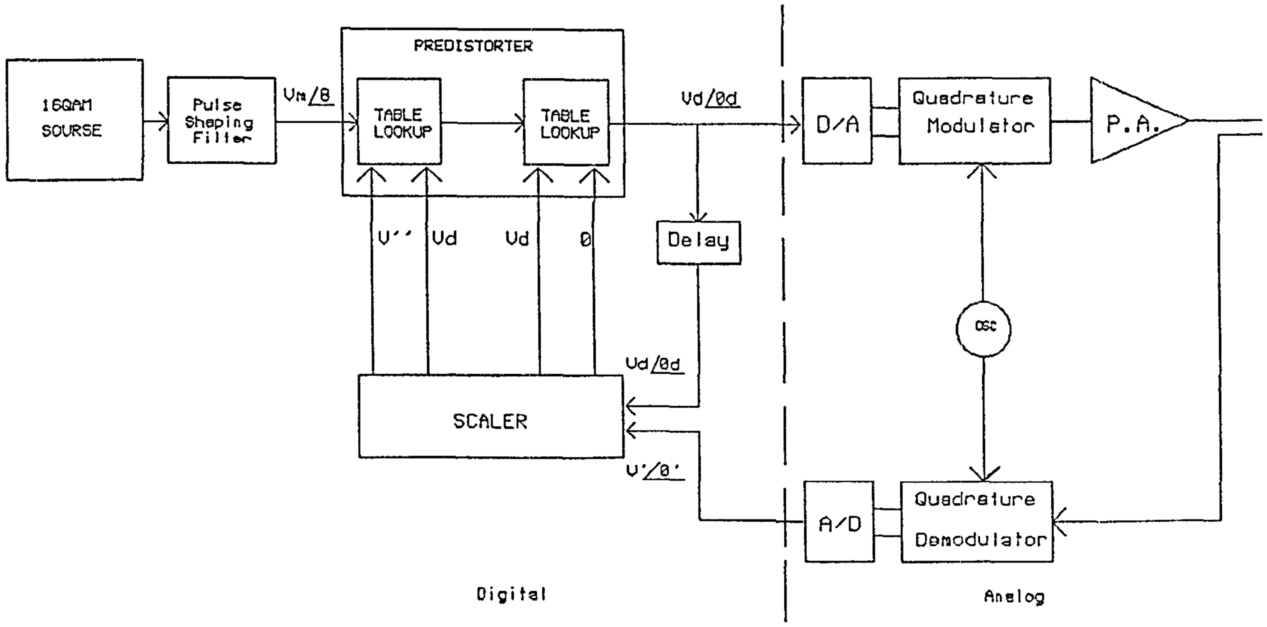


Fig. 1: Adaptive predistorter diagram

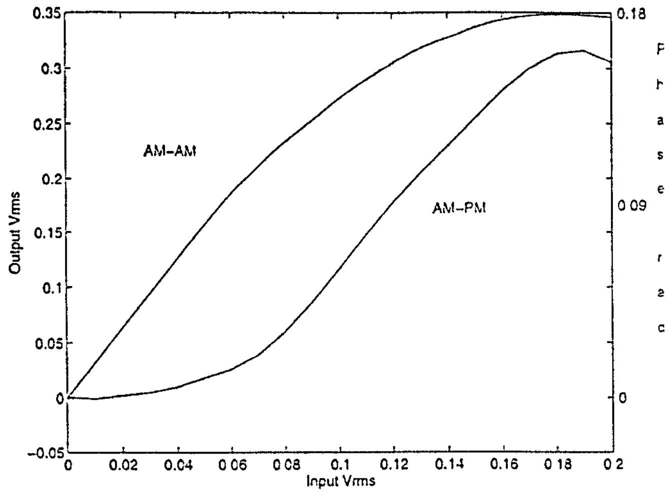


Fig.2· Power amplifier characteristics. Class AB

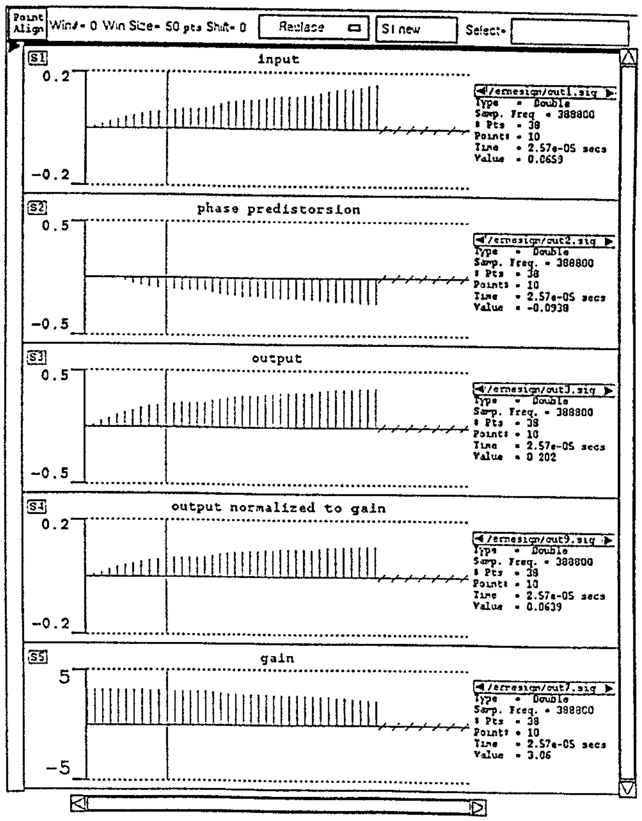


Fig. 3· Representation of 38 semples of: S1) Input signal, S2) Phase predistortion, S3) Output signal, S4) Output normalized to gain K, S5) Gain

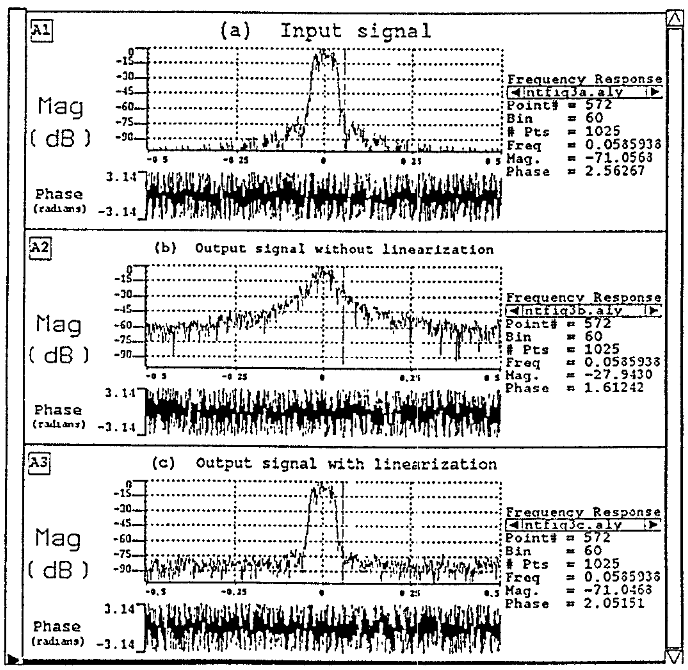


Fig 4· Power spectral density of: a) Input signal, b) output signal without linearization, c) output signal with linearization