Nonlinear Amplifier Modeling Taking Into Account HF Memory Frequency

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Abstract — **A model to describe the nonlinear behavior of a power amplifier is proposed, which extends the power series expansion to take into account frequency dispersion of a single tone and a two-tone input signal. Behavorial model is commonly estimated according to CW measurements but, even if this model is accurate for one-tone signal, no predictions can be done for intermodulation products of a N-tones signal.**
This paper focuses on a nonlinear behavorial model which is divided in a dynamic gain and a balance filter. The dynamic gain consists in a nonlinear digital filter and a complex polynomial behavorial representation, the latter being extracted from a referenced frequency AM-AM and AM-PM characterization. The balance filter consists in a nonlinear generator, whose aim is to compensate phase and gain intermodulation products which are simulated by the dynamic gain. Our main contribution is to propose a new method to assess the model of the amplifier which can be extended to UMTS signals.

I. INTRODUCTION

The main concern in a wireless communications system is the efficiency and the linearity of the power amplifiers (PA). The more efficient is the power amplifier, the smaller is the battery. Besides, this entails a cheaper cooling system and extends speech time. Generally, a power amplifier is efficient when input signal is driven near the 1dB compression point. However, this will result in distorsion and intermodulation. Unless the signal is robust with respect to nonlinearities (like modulation with constant envelope such as Minimum Shift Keying MSK), its bandwidth is widened by odd order non linearity (spectral regrowth). With amplitude envelope fluctuation, AM-AM and AM-PM distorsions are dispatched towards the output signal.

To estimate the nonlinear effects, several methods have already been developed : Volterra Series, differential equations, harmonic balance, transient simulation and behavorial model [1]. However, only the behavorial model can be used easily in a digital communication system simulation. Recently, some new statistical techniques were developed to analyze the effects of non linear distortion on a CDMA signal (used in the next generation telecommunications standard), and also to predict ACPR (Adjacent Channel Power Ratio) from the AM-AM and AM-PM characteristics [2,3]. Nevertheless, such models assume that gain and phase characteristics are constant over the entire band frequency of the input signal, which is only true for narrowband signal. Other models have been suggested for more accurate simulations of wideband signal [4], but none of them seems to provide an efficient implementation with easily measured parameters. A new method based on time domain measurement technique [5] has been proposed to model the memory effects. An enhancement is achieved for pulse input, but since the ARMA filter coefficients do not depend on input amplitude, differences occur for other input signals.

II. MODEL DESCRIPTION

Usually, AM-AM and AM-PM measurements are realized in the band center of the amplifier. As to a memoryless behavorial model, the amplifier is represented by a complex polynomial series, whose coefficients are either estimated with the mean squares method according to CW measurements, or assessed with the amplifier characteristics such as the gain, the compression point and the two-tone third order intercept point (IP3). Even if this last method does not give any information about the phase (AM-PM), it is a commonly used method. The relationship between the characteristics of the amplifier, (gain G, 1dB compression point P1dB, IP3) given in the data book and the real coefficients (no phase is taken into account) of the polynomial model are given in [6].
Whatever the chosen methods, the output signal is approximated with the following baseband input/output relationship

$$e_o(t) = e_i(t)\sum_{n=0}^{N} a_{2n+1}\left|e_i(t)\right|^{2n} \tag{1}$$

where a_{2k+1} is a complex power series coefficient and $e_i(t)$ is the complex input translated in baseband [3].

Knowing G, P1dB and IP3, we can model the amplifier by a fifth order power series.
Nevertheless, a better model can be achieved in a context of real polarisation and loading conditions with AM-AM and AM-PM measurements according to the least squares method. This method consists in approximating the output versus the input power and the phase difference between the output and the input signal versus the input power via a complex polynomial.

However, even if such memoryless model leads to very good results for narrowband and constant envelope signal like CW or GMSK, differences are expected at another frequency and no agreement is achieved at the intermodulation products for other signals (QPSK, QAM, ...). As a result, it is necessary to develop a model which is valid in a large frequency band and which can simulate with accuracy the spectral regrowth.

III. PROPOSED MODEL

In this paper, we aim at improving the model of the amplifier, taking into account HF frequency dispersion for a one-tone and a two-tone input signal.
First, we propose a model to take into account the HF frequency dispersion of the gain. Then, we will improve the model for a two-tone input signal.

A Dynamic Gain

To model the dynamic gain, a test bench has specially been developed in order to determine the AM/AM and AM/PM characteristics of a microwave amplifier on a large bandwidth. For example, figure 1 presents the gain compression characteristics for ZJL-3G Alcatel amplifier. In figure 1, we can observe that the gain and the distorsions depend on frequency.

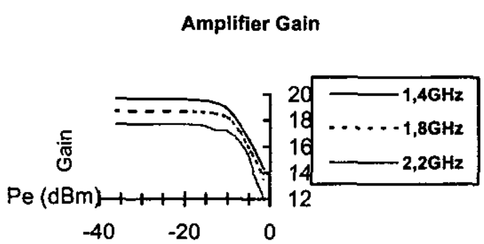

Figure 1 : Gain versus input power measured at different frequencies

To take into account the frequency dispersion, the idea is to develop an *auto-regressive moving average* (ARMA) or *moving average* (MA) filter, which is defined in a frequency domain. This filter aims at compensating attenuation and phase difference between the different gain compressions and phase distorsions measured for different frequencies according to a reference curve. For stability reasons, MA filter is preferred to ARMA filter, except unusual amplifier which presents high variations of the gain versus frequency.

As a result, the amplifier is defined as follows :

$$G(\tilde{z}) = \sum a_n \tilde{z}^n \text{ (nonlinear model)} \quad (2a)$$

$$\text{and } H(f) = \sum_{k=0}^{P-1} a_k(V_e) e^{j2\pi kf/P} \text{ (filter model)} \quad (2b)$$

First, we assess the coefficients of the nonlinear model at a reference frequency by the least squares method to fit the AM-AM and AM-PM characteristics. At this reference frequency, the filter attenuation is unitary and the model is similar to the memoryless model.
Then, the coefficients of the filter are estimated by solving the following equation with the least-squares method :

$$\sum_{k=0}^{R-1} a_k(Ve) e^{j2\pi f_j T_e k} = \frac{V_e(f_{ref})}{V_e(f_j)}$$

where Te is the sampling period. The number of equations is less or equal to the number of frequencies used to measure the AM-AM and AM-PM characteristics. The precision of the model can be improved by making more measurements at other frequencies, except if the matrix is ill-conditioned. This can occur if too many coefficients are used to model the filter, or if too many measurements are made to estimate the coefficients of the filter.

As the filter is determined by step of Ve, the filter is said to be nonlinear. The advantage of digital filter lies in the relationship between Z transform and temporal domain. Indeed, if we sample x(t) at time step Te, the input-output relation of FIR filter can be expressed as :

$$y(nTe) = \sum_{p=0}^{L-1} \alpha_p x[(n-p)Te] \quad (3)$$

where x(nTe) and y(nTe) denote the complex nth sampled input and output of the filter at time nTe. In this way, HF memory effect can be taken into account.

The reference curve is usually chosen at the reference frequency of the modulated signal in order to convert downwards modulated signal, into baseband and to evaluate filter coefficients. Figure 2 shows very good agreement between the proposed model (crossed) simulation for an AM-AM characterization and the measurements (solid line) compared with the memoryless model (dotted line). The memoryless model has been characterized at 1.515GHz and corresponds to the nonlinear model G (2a). The input signal is a CW signal at 1.6GHz for different input power levels.

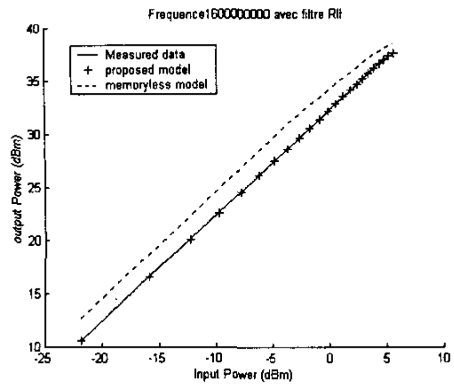

Figure 2 :Comparison between the measured data, the dynamic gain model and a memoryless model.

The proposed model is particularly appropriate for GSM communication system because the envelope of the GMSK signal is constant and the gain of the amplifier varies between the different carriers frequencies.
Nevertheless, we aim at adapting the amplifier to two-tone signal.

B Frequency generator

Indeed, even if the dynamic gain model is accurate for the single tone signal (figure 2), the model is not sufficient to simulate intermodulation product for a two-tone input signal. In fact, the proposed model produces undesirable magnitude at the intermodulation products. The idea is to add a nonlinear frequency generator to compensate the phase and gain difference between the intermodulation products simulated and the measurements. A new test bench has been developed to measure with accuracy the amplitude and the phase of the fundamentals carrier and the intermodulation products of an amplifier driven by a two-tone input signal with different tones spacing.
The model can be represented as follows (figure3).

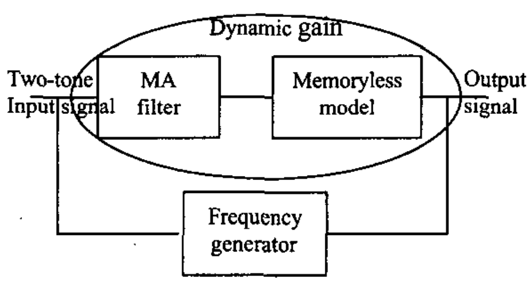

Figure 3 : Two-tone proposed model

The new model looks like order 2 Volterra series, where frequency generator can be compared to $H_2(f)$. Nevertheless, the way of computing the frequency generator is completely different from the estimation of Volterra kernel.
Indeed, the function of the frequency generator is also estimated by the least squares method in frequency domain. Thanks to the test bench, we measure the power at the output intermodulations products $(2f_1 - f_2)$ at different input power level and for different tones spacing. The amplifier is driven by a two-tone signal where the first frequency is f_1 and the second is swept around f_1. This kind of measurement point out the effects of BF frequency dispersion when the difference frequency $f_1 - f_2$ is small. The output power measured at $2f_1 - f_2$ is substracted to the dynamic gain simulated output power.
The frequency generator is modeled as a nonlinear filter

$$H_2(f) = \sum_{k=0}^{R-1} b_k(V_e) e^{j2\pi fk/R}$$

At a fixed input power, we assess the coefficients of the filter by solving the following equation with the least squares method :

$$\sum_{k=0}^{R-1} b_k(Ve) e^{j2\pi f_1 T_e k} = \frac{V_s(2f_1 - f_2)}{V_e(f_1)}$$

To validate the proposed model, figure 4 compares the measured and simulated intermodulation products $2f_1 - f_2$ at specific frequencies and input power of the two-tone input signal.

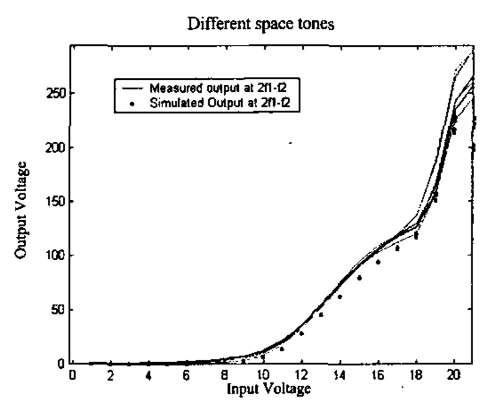

Figure 4 : Output simulated and measured power at the intermodulation products for different distance of the tones (colour) versus input voltage.

To complete the test of the suggested model, we compare in figure 5, C/I3 measured and simulated for different two-tone signals, whose frequency and input power is different from data used to assess the model. The maximum average power of the input signal is taken at 2 dB under the 1dB point compression.

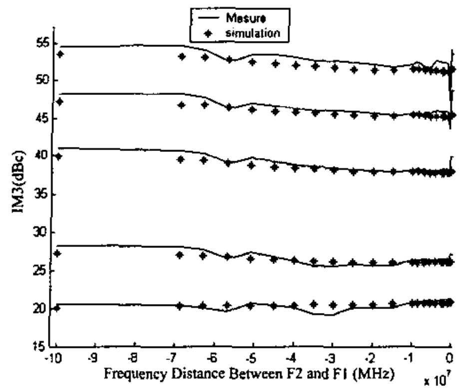

Figure 4 : C/I3 at different input power and frequency .

IV. Conclusion

The model that we propose is based on AM-AM and AM-PM characteristics, which is used to model the dynamic gain. Good results are achieved for any CW signal at any input power and any frequency. The balance filter is extracted from two-tone measurements. The error did not exceed 5dB between the simulation and the measured C/I3 data when the amplifier is driven by an input two-tone signal whose frequency and input power is different from the data used to model the amplifier. In fact, cubic interpolation function used in the proposed model is not the most appropriate interpolation function to the problem. One solution would be to substitute the generator function by a 3rd order polynomial model and a filter. We are currently working on a best interpolation function and on the Hammerstein model to model UMTS signal.

References

[1] J.F. Sevic, M.B. Steer, A.M. Pavio, "Nonlinear Analysis Methods for the simulation of digital wireless communication systems", International Journal of Microwave and Millimeter-Wave Computer Aided Engineering, Vol. 6, No. 3,pp 197-216, Jan 1996.

[2] S.W. Chen, W. Panton, R. Gilmore, "Effetcs of nonlinear distorsion on CDMA communication system", IEEE Transactions on Microwave Theory and Technique, vol. 44, No. 12, pp 2743-2749, Dec 96.

[3] K.G. Gard, H.M. Gutierrez, M.B. Steer, "Characterization of spectral regrowth in microwave amplifiers based on the nonlinear transformation of a complex gaussien process", IEEE Transactions on Microwave Theory and Technique, vol. 47 No. 7, pp 1059- 1069, Jul 97.

[4] M.C. Jeruchim, P. Balaban, K.S. Shanmugan, "Simulation of communication system", New York, Plenum, 1992.

[5] C..J. Clark, G. Chrisikos, M. Muha, A.A. Moulthrop, C.P. Silva, " Time-domain envelope meaurement technique with application to wideband power amplifier modeling", IEEE Transactions on Microwave Theory and Technique, vol. 46, No. 12, pp2531-2539, Dec 98.

[6] L.Lecheminoux, M.Villegas, " ACPR to Third Order Interception Point and 1 dB Compression Point New Relation", 2000 European Conference on Wireless Technology, Paris