Mixed time and frequency-domain identification of Radio Frequency Power Amplifiers

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Abstract—**This paper presents a modeling and identification procedure for Radio Frequency Power Amplifier (PA) in continuous time representation. The proposed method is based on a combined approach using time and frequency domain. The microwave PA dynamics are described by the continuous Hammerstein model. The distortion function coefficients were estimated using Least Mean Squares (LMS) method that minimizes the squared error based on baseband output data. The coefficients of dynamic block are extracted using an iterative instrumental variable with measured PA frequency responses. To demonstrate the feasibility of the identification method in experimental results we used a standard mobile PA at 1.85GHz.**

I. INTRODUCTION

The microwave power amplifier (PA) represents one of the major element of modern wireless communication systems. The modeling and simulation of PA nonlinearities and memory effects, in particular their impacts in a transmission digital system, are a current topic of intensive research worldwide [1][2]. Numerous approaches in PA identification area have been developed to characterize the input to output complex envelope relationship [3][4].

Block-structured models are used to model Radio Frequency (RF) PA system that can be represented by interconnections of linear dynamics and static nonlinear elements [1]. Hammerstein model is one of these cascaded models that composed from static nonlinearity followed by a linear dynamic system [5]. In literature, the existing identification methods of Hammerstein model can be divided into time-domain iterative methods, stochastic methods separable least squares methods and frequency domain methods [6]. In the proposed method, a mixed time and frequency domain identification approach was considered to analyze baseband data from microwave amplifier. The ultimate aim is the estimation of the PA model parameters by minimizing the quadratic estimation error based on improved time-delay data for nonlinear function and frequency response for the memory effect of the Hammerstein model.

For frequency-domain identification, gain and phase-shift measurements are used commonly to construct a quadratic criterion that is minimized by ordinary LMS algorithm [7]. The main problem in these techniques is assuring the optimal convergence under noisy sampled data [7][8]. To guarantee robustness and consistent estimates in presence of output disturbances, we introduce the Instrumental Variables (IV) on the identification procedure. This frequency-domain procedure with IV estimator is the major contribution of this paper.

Two experimental setups dedicated to radio frequency with baseband signal transmission were performed and used in order to validate this technique. The results confirm the PA characterization accuracy using mixed time/frequency representation.

II. PA MODEL DESCRIPTION

The PA behaviour is modeled by a continuous Hammerstein model operating on baseband quadrature I/Q time-domain waveforms. In this model, the complex low-pass equivalent representation of the communication signal is used to avoid the high sampling rate required at the carrier frequency. As shown in Fig. 1, this model consists of static nonlinearity followed by a multivariable Linear Time Invariant (LTI) system.
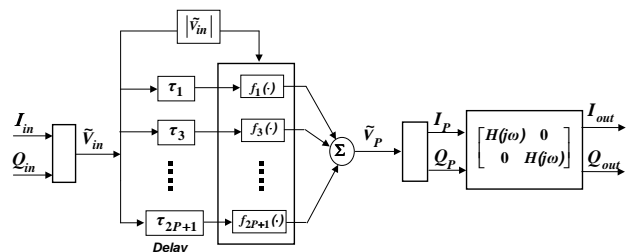

Fig. 1. Radio frequency power amplifier model

The nonlinear distortion of the amplifier usually results from nonlinear distortion processes in the transistors that make up the amplifier. In the model, the static distortion function gives a complex envelope $\widetilde{V}_P = I_P + j.Q_P$ of the transmitted input signal $\widetilde{V}_{in} = I_{in} + j.Q_{in}$ according to the polynomial function composed by even terms which produces harmonic distortions inside the PA bandwidth. A delay is added for each order to improve nonlinear memory modeling:

$$\widetilde{V}_{P_k} = \sum_{p=0}^{P} \widetilde{c}_{2p+1} \cdot |\widetilde{V}_{in_k}|^{2p}.\widetilde{V}_{in_{k-\tau_{2p+1}}} \qquad (1)$$

where $\widetilde{c}_{2p+1}$ are the complex coefficients of the power series, τ_{2p+1} are the delays for each component; $\widetilde{V}_{in_k}$ and $\widetilde{V}_{P_k}$ are the sampled input and output complex envelopes of the nonlinear static model.

The dynamical model including memory effects caused by the PA can be expressed with a differential equation. As shown in Fig. 1, the input $\widetilde{V}_P$ to output $\widetilde{V}_{out} = I_{out} + j\,Q_{out}$ relation of this n^{th} order filter can be written as:

$$\widetilde{H}(j\omega) = \frac{I_{out}}{I_P} = \frac{Q_{out}}{Q_P} = \frac{1 + \sum_{i=1}^{m} b_i.(j\omega)^i}{\sum_{i=0}^{n} a_i.(j\omega)^i} \quad (2)$$

where the coefficients $\{a_k\}$ and $\{b_k\}$ are real scalars that define the model.

Our objective is to identify the model parameters according to time domain and frequency response of the amplifier. Thus, we define the transposed parameters vector:

$$\underline{\theta} = [\underbrace{a_0 \cdots a_{n-1}\ b_1 \cdots b_m}_{\theta_F}\ \underbrace{\widetilde{c}_1 \cdots \widetilde{c}_{2P+1}}_{\theta_P}]^T \quad (3)$$

Note that $\underline{\theta}$ will be determined as a combination of two subvectors; $\underline{\theta}_P$ describing the memoryless function and $\underline{\theta}_F$ corresponding to the filter parameters. In the next sections we present an identification procedure based on two steps:

- identification of $\underline{\theta}_P$ using the two-tone test and Least Mean Squares algorithm (LMS),
- identification of $\underline{\theta}_F$ using a new procedure based on frequency gain and phase responses and unbiased LMS method with IV estimator.

A. Memoryless function identification

The identification principle of nonlinear parameter $\underline{\theta}_P$, using complex envelope waveforms, is shown in Fig. 2.
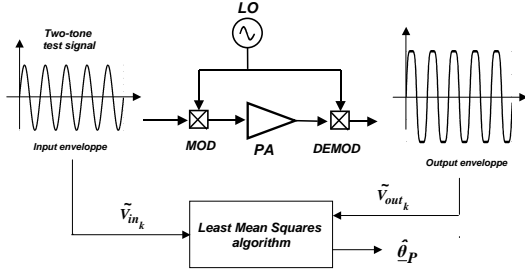

Fig. 2. PA identification scheme for static analysis

The PA characteristics are performed by a two-tone test signal applied on baseband inputs I_{in} and Q_{in} at fixed low frequency and high input level. In these conditions, the PA filtering effects are assumed negligible according to nonlinear dynamics. To applied a LMS algorithm, it is necessary to write the relation (1) as a linear regression according to the parameters $\widetilde{c}_{2p+1}$ such as:

$$\widetilde{V}_{P_k} = \underline{\varphi}_k^T \cdot \underline{\theta}_P \quad (4)$$

with $\underline{\varphi}_k = [\widetilde{V}_{in_{k-\tau_1}}\ |\widetilde{V}_{in_k}|^2 \widetilde{V}_{in_{k-\tau_3}}\ \cdots\ |\widetilde{V}_{in_k}|^{2P} \widetilde{V}_{in_{k-\tau_{2P+1}}}]^T$ is the regressor vector.

For a time-domain acquisition with K data of input complex envelope $\widetilde{V}_{in}$ and disturbed output noted $\widetilde{V}_{out}^*$, a solution for the coefficients is obtained by minimizing the normalized mean-squared error with LMS algorithm [7] such as:

$$\hat{\underline{\theta}}_P = \left(\sum_{k=1}^{K} \underline{\varphi}_k \cdot \underline{\varphi}_k^H\right)^{-1} \cdot \sum_{k=1}^{K} \underline{\varphi}_k \cdot \widetilde{V}_{out}^* \quad (5)$$

where :

$(.)^H$ denotes transpose-conjugate transformation

Noted that vector estimate $\hat{\underline{\theta}}_P$ is asymptotically unbiased because the regressors vectors $\underline{\varphi}_k$ are not correlated with the measured output $\widetilde{V}_{out}^*$ [9].

B. Filter identification using frequency response

Fig. 3 shows the principle of filter parameters $\underline{\theta}_F$ identification from PA frequency response data using LMS algorithm with iterative IV estimator.
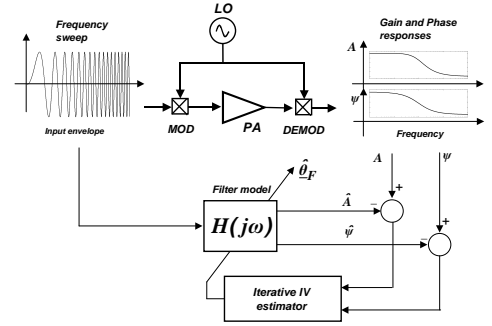

Fig. 3. PA identification scheme for dynamic identification

Assume that we have measured *Bode* diagram of the transmission system which correspond to gain A_k and phase-shift ψ_k for K values of frequency ω_k. At any frequency ω_k, we can write:

$$\frac{1 + \sum_{i=1}^{m} b_i.(j\omega_k)^i}{\sum_{i=0}^{n} a_i.(j\omega_k)^i} = \widetilde{G}_{\omega_k} \quad (6)$$

where $\widetilde{G}_{\omega_k} = A_k \cdot e^{j\,\psi_k}$

We can rewrite the above equation as:

$$\sum_{i=0}^{n} a_i.\widetilde{G}_{\omega_k}.(j\omega_k)^i - \sum_{i=1}^{m} b_i.(j\omega_k)^i = 1 \quad (7)$$

or into linear regression version:

$$\underline{\varphi}_k^T \cdot \underline{\theta}_F = 1 \quad (8)$$

with $\underline{\varphi}_k = [\widetilde{G}_{\omega_k} \cdots \widetilde{G}_{\omega_k}.(j\omega_k)^n\ -(j\omega_k) \cdots -(j\omega_k)^m]^T$

The suggested method shown in Fig. 3 minimizes the following quadratic criterion:

$$J = \sum_{k=1}^{K} \left| \sum_{i=0}^{n} a_i . \widetilde{G}_{\omega_k} . (j\omega_k)^i - \sum_{i=1}^{m} b_i . (j\omega_k)^i - 1 \right|^2 \quad (9)$$

Because a_i and b_i are non-complex parameters, it is necessary to construct a new regressors vectors with scalars coefficients. Splitting $\underline{\varphi}_k$ into real and imaginary parts $\underline{\varphi}_{R_k}$ ans $\underline{\varphi}_{I_k}$ respectively, and then equating the real and imaginary parts on both sides of the relation (8):

$$\begin{bmatrix} \underline{\varphi}_{R_k}^T \\ \underline{\varphi}_{I_k}^T \end{bmatrix} . \underline{\theta}_F = \begin{bmatrix} 1 \\ 0 \end{bmatrix} \quad (10)$$

For the K pulsations ω_k, we obtain K sets of the above relation and we can write a global system:

$$\phi . \hat{\underline{\theta}}_F = \underline{Y} \quad (11)$$

where $\underline{Y} = [\,1 \,\cdots\, 1\, 0 \,\cdots\, 0\,]^T$ and ϕ is a regression matrix composed by $\underline{\varphi}_{R_k}$ and $\underline{\varphi}_{I_k}$.

Then, the least-squares estimate noted $\hat{\underline{\theta}}_{F_{LS}}$ is given by:

$$\hat{\underline{\theta}}_{F_{LS}} = (\phi^T \phi)^{-1} \phi^T \underline{Y} \quad (12)$$

When the gain and phase-shift data are disturbed, which is the case on experimental tests, the LMS algorithm in the above relation gives biased and inconsistent estimates because the regressors are correlated with the output measurements. A common strategy for dealing with this bias problem is to use Instrumental Variable (IV) estimator [7]. The principal idea of the IV method is to premultiply the regression matrix ϕ in (11) with a suitable matrix $Z = [\underline{z}_1 \, \underline{z}_2 \, \cdots \, \underline{z}_K]$ whose elements $\underline{z}_k$ are uncorrelated with output disturbances. This gives a new estimate parameters called *IV* estimates such as:

$$\hat{\underline{\theta}}_{F_{IV}} = (Z^T \phi)^{-1} Z^T \underline{Y} \quad (13)$$

The elements of the regression matrix Z are called the instruments or 'instrumentals variables'. These elements are usually formed as delayed and filtered values of the inputs and/or the outputs and have the same structure as the regressors vectors $\underline{\varphi}_k$. An original choice is based on the use of the estimated parameters $\hat{\underline{\theta}}_F$ for simulation of frequency response $\hat{H}(j\omega)$ and generate estimates gain $\hat{A}$ and phase-shift $\hat{\psi}$. With these uncorrelated variables, we construct the matrix Z defined by a new regressor vector :

$$\underline{z}_k = [\,\hat{G}_{\omega_k} \, \cdots \, \hat{G}_{\omega_k} . (j\omega_k)^n \,\, -(j\omega_k) \, \cdots \, -(j\omega_k)^m\,]^T \quad (14)$$

where $\hat{G}_{\omega_k} = \hat{A}_k . e^{j\hat{\psi}_k}$

Noted that each regressor vector $\underline{z}_k$ is associated with a prospected frequency ω_k. This method can be used iteratively with a parallel model initialized with LMS estimates $\hat{\underline{\theta}}_{F_{LS}}$.

III. EXPERIMENTAL RESULTS

In this section, the LMS and IV algorithms presented previously are used to identify the nonlinearity function and transfer function model of a standard amplifier for mobile communication.

A. PA setups and characteristics

The power amplifier is a commercial APM1855 from AD-VANCED SEMICONDUCTOR BUSINESS INC. The quadrature modulator AD8349 and demodulator AD8347 are inserted at the input and output of the PA. They are standard commercial units from Analog Devices. The local oscillator frequency is 1.85GHz obtained from Digital Modulation Signal Generator (Anritsu MG 3660A). Two measurement setups, shown in Fig. 4, are used:

(a) the left figure (4.a) illustrates time-domain data acquisition. The two-tone signal at low frequency equal to 5 kHz are delivered by a Tektronix AFG3102 Arbitrary function Generator connected to PC control. Input and output data are obtained from YOKOGAWA Digital Oscilloscope with a sampling period equal to 10ns.

(b) figure (4.b) shows the test bench used to obtain the frequency response. A network analyzer generates the frequency sweep signal and measures the frequency transfer function.
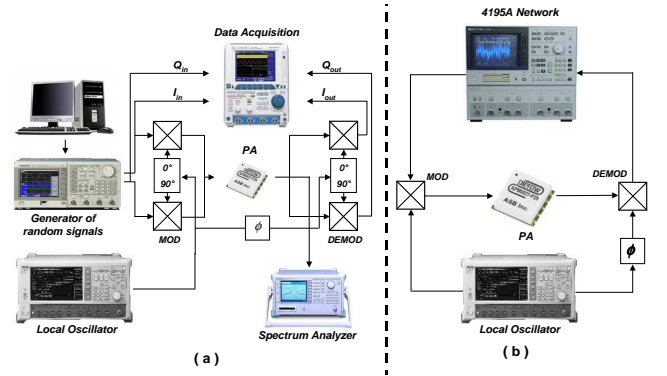

Fig. 4. PA setups for (a) time-domain analysis (b) frequency-domain analysis.

Fig. 5 illustrates nonlinear characteristics appearing as amplitude to amplitude distortion (AM/AM) and amplitude to phase distortion (AM/PM).
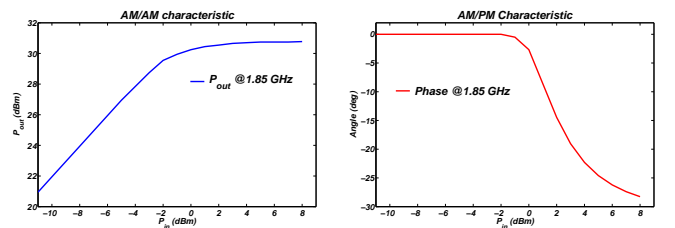

Fig. 5. AM/AM and AM/PM characteristics of used PA

B. Identification results

All estimation results presented here were produced using 20 Monte-Carlo acquisitions and the informations on the model parameters and figures are based on the mean of those values.

For nonlinear function identification, a PA is driven in saturation with a two-tone test at frequencies $f_{1,2} = 1850 \pm$

5kHz. Fig.6 shows a measured I channel normalized output waveform and its approximation with a 5th order ($P = 2$) polynomial function:

$$\widetilde{V}_{P_k} = \widetilde{c}_1.\widetilde{V}_{in_{k-\tau_1}} + \widetilde{c}_3.|\widetilde{V}_{in_k}|^2\widetilde{V}_{in_{k-\tau_3}} + \widetilde{c}_5.|\widetilde{V}_{in_k}|^4\widetilde{V}_{in_{k-\tau_5}}$$
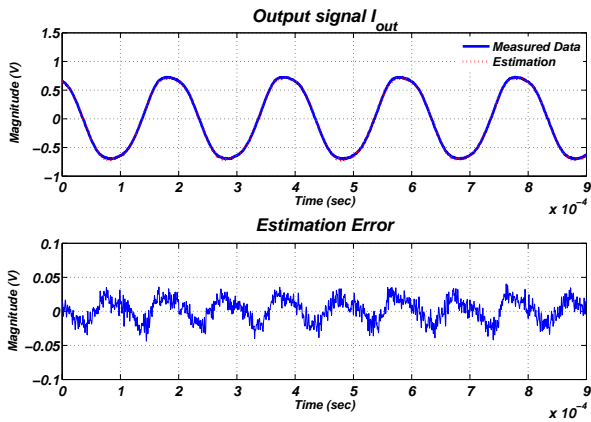

Fig. 6.   Comparison between measured waveform and estimation

For the 20 realizations with normalized data, LMS algorithm described in relation (5) gives:

$$\begin{cases} \widetilde{c}_1 = 1,2178 - j\,4,052.10^{-2} \\ \widetilde{c}_2 = -2,144.10^{-1} + j\,2,453.10^{-2} \\ \widetilde{c}_3 = -7,678.10^{-2} - j\,1,631.10^{-2} \end{cases}$$

with $\tau_1 = 6$, $\tau_3 = 45$ and $\tau_5 = 120$ samples delay.

The estimated filter is a 3th order transfer function with one zero ($n = 3$ poles and $m = 1$ zero) corresponding to the minimum value of the cost function defined by the quadratic criterion (Eq. 9) after identification. Fig. 7 shows the gain and phase-shift responses for 400 equally distributed frequencies on the logarithmic scale in the band [160Hz – 16MHz]. As it can be seen, the model interpolates both sources of data with a negligible error.
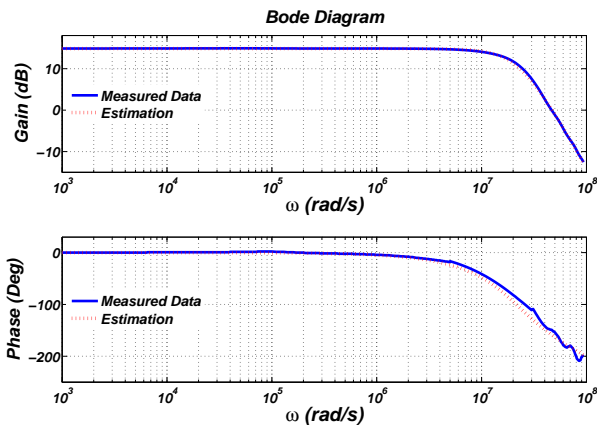

Fig. 7.   Comparison between measurement data and estimation

The IV algorithm gives the following filter parameters:

$$\begin{cases} a_0 = 1,809.10^{-1} \\ a_1 = 1,483.10^{-8} \\ a_2 = 5,031.10^{-16} \\ a_3 = 3,170.10^{-24} \\ b_1 = -1,952.10^{-10} \end{cases} \quad (15)$$

The achieved transfer function corresponds to a Low-pass filter with a cut-off frequency in baseband format at 1,6 MHz.

IV. CONCLUSIONS

In this paper, the solution for time and frequency domain estimation of RF PA continuous model is presented. A proposed identification method, even based on Hammerstein model and mixed time and frequency analysis can easily put into practice, especially for RF transmission system. The central result of this paper was the implementation of a method using unbiased LMS algorithm with IV estimator for the identification of LTI systems.

Consistency and efficiency of IV estimator are important properties to ensure the robustness of parameter convergence in presence of noisy data. The experimental result shows the application of these class of characterizations methods that combines both frequency and time domain data.

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