AN ADAPTIVE FEEDFORWARD AMPLIFIER APPLICATION FOR 5.8 GHZ

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ABSTRACT

In this study, a 5.8 GHz power amplifier is linearized by adaptive feedforward technique. A DSP based control scheme is applied to reduce the intermodulation distortion of this amplifier. Two-tone test is used to verify the design.

I. INTRODUCTION

All radio transmitters contain nonlinear components, such as amplifiers and mixers. If the transmitted signal has a fluctuating envelope, such as in linear data modulations QPSK or 16QM, or in multi-carrier configurations at a cellular base station, then intermodulation (IM) distortion is generated in the nonlinear components. Since, most of the IM power appears as interference in neighboring channels, it is necessary to use highly linear amplifiers for those applications.

Several linearization approaches have so far been developed. Predistortion technique has the advantage of unconditional stability but, it has limited accuracy when implemented with the analog technology. With digital technology, its bandwidth is limited to one or two cellular channels by the computational rate. Feedback linearization is simple but, it reduces the gain and the stability considerations limit its bandwidth and accuracy. A third category which is named as feed-forward linearization has been studied some times. A feedforward linearizer has some distinct advantages. Since the signals are manipulated by inherently wideband analog technology it can handle multicarrier signals at a mobile base station. Also, it is nonparametric; that is, it does not rely on any knowledge of the signal structure or any family of curves, such as polynomials, to represent the amplifier characteristic. Unfortunately, it is based on subtraction of nearly equal quantities therefore, it is sensitive to component tolerances and drift and also to the change in power level when the number of carriers changes. However, the recent development of adaptation methods to compensate these effects and other changes has led to renewed interest in feedforward linearization [1].

Feedforward is the most effective and broadly used linearization technique employed in modern multi-carrier and digital communication systems. The key operating parameters are adjusted by means of some mechanism of automatic adaptation. The key aspects of the feedforward linearization technique are the amplitude and phase imbalances as well as the inequality of the signal delays among the different branches which are compared [2].

In this study a brief description of typical feedforward architecture is made and the simulation results of a adaptation technique for these imbalances which are found by using the least mean square (LMS) algorithm. Simulation is done entirely in the Agilent's ADS simulation tool and the results are presented.

II. FEEDFORWARD TECHNIQUE

The architecture of the well-known feedforward linearizer has the form as illustrated in Fig.-1. Since the power amplifier (PA) presents amplitude and phase distortion, it will be assumed that its output is made up of an amplified version of the input signal plus certain intermodulation (IMD) products. A feedforward amplifier achieves its linearity improvement by canceling the intermodulation products produced by the main amplifier. A signal composed of only the distortion produced by the main amplifier is generated by the carrier cancellation loop. The error signal is adjusted to be equal in amplitude but 180^0 out of phase with the IMD products of the main amplifier. By adding the error signal of the main amplifier, IMD cancellation is achieved. Ideally, the linearization technique of the amplifier aims to eliminate completely the distortion present in the PA output signal.

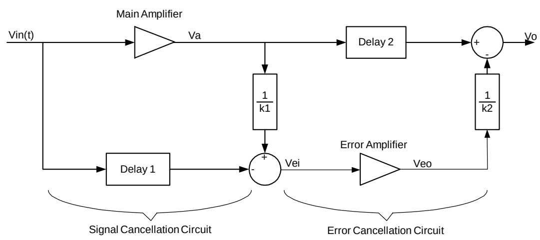

Fig. 1. General Architecture of Feedforward Amplifier

A feedforward amplifier consists of two loops. The first loop is a carrier cancellation loop, which is used to cancel the carrier and obtain the IMD products of the main amplifier (denoted as the error signal). The second loop is the IMD cancellation loop, which used to reduce the output IMD products with the error signal. The amount of correction is limited by the ability of the two loops to match gain and phase between the main signal and error paths.

III. MATHEMATICAL REPRESENTATION

For simulation purposes the equivalent adaptive feedforward amplifier model is shown in Figure 2. The PA and the error amplifier are represented by the complex gains G and g, respectively. While G is a function of the input signal amplitude, g is assumed to be constant which implies a linear operation of the error amplifier. The same power amplifier can be in both loops. For the first loop it is used as power amplifier because of high power operations. On the other hand, since the second loop operates in much lower power levels it is used as linear error amplifier in this loop. The terms ($1/k_1$) and ($1/k_2$) represent the coupling factors of the directional couplers used in the signal cancellation circuit and the error cancellation circuit, respectively. The complex quantities a and b constitute adjustable parameters for the compensation of the gain and phase imbalances among the branches that are compared.

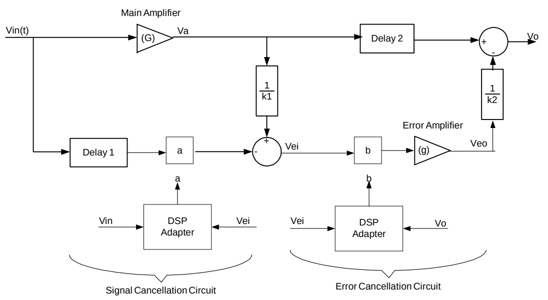

Fig. 2. Adaptive Feedforward Amplifier Model

Input signal can be assumed as;

$$V_{in} = ce^{j\phi(t)} \tag{1}$$

As previously mentioned, the nonlinear PA introduces amplitude and phase distortion. The PA output is

$$V_a = G_1 V_{in} + v_{IMD} \tag{2}$$

$G_1 = |G_1| e^{j\alpha}$: The linear gain of PA

$G_1 v_{in}$: Amplified version of input signal plus a certain phase shift α

v_{IMD} : Intermodulation products

In the signal cancellation circuit a fraction of the PA output signal v_a/k_1, with k_1 real, is compared with a sample of the input av_{in}, with a complex, resulting in an error signal v_{ei} given by

$$v_{ei} = \frac{G_1}{k_1} v_{in} + \frac{1}{k_1} v_{IMD} - av_{in} \tag{3}$$

If a is adjusted in such way that

$$a = \frac{G_1}{k_1} \tag{4}$$

The error signal v_{ei} will contain only the intermodulation products.

$$v_{ei} = \frac{1}{k_1} v_{IMD} \tag{5}$$

In order to adjust these gain and phase imbalance accurately, the complex parameter a will be altered by means of an adaptive procedure. In a similar way, in the error canceling circuit, the error signal is amplified in a second(error) to obtain;

$$v_{eo} = g v_{ei} \tag{6}$$

Where, $g = |g| e^{j\beta}$ is the gain of the second amplifier. In a way similar to the previous case, the gain and phase of the signal v_a will be adjusted by means of the complex parameter b, before comparing it to the signal v_o free of distortions

$$v_o = V_a - \frac{v_{eo}}{k_2} \tag{7}$$

where k_2 is a real parameter. Using (2), (3) and (6) in (7) the output voltage can be formed as follows;

$$v_o = G_1 v_{in} + v_{IMD} - b\frac{g}{k_1 k_2} v_{IMD} \tag{8}$$

If b is adjusted such that,

$$b = \frac{k_1 k_2}{g} \tag{9}$$

The equation for v_o becomes

$$v_o = G v_{in} \tag{10}$$

Which is simply equel to the linear gain of the power amplifier. The complex parameter b will be adjusted by means of an adaptive algorithm similar to the one used with the parameter a.

IV. ADAPTIVE SOLUTION

An effective solution for an adaptive estimate of the complex parameters a and b includes the use of digital signal processing (DSP) in both cancellation circuits.

The complex parameters a and b will be adjusted according to the LMS (Least Mean Square) algorithm implemented in the numerical domain (Help of Agilent ADS simulator). Only the equations corresponding to the adaptation of parameter a are developed with the assumption that the same ones are completely applicable to the parameter b.

According to the LMS algorithm, the equations for a are given by

$$a(n) = a(n-1) + \Delta \cdot v_{in}(n) \cdot v_{ei}^{*}(n) \qquad (11)$$

$$a(n) - a(n-1) = K \cdot \Delta\tau \cdot v_{in}(n) \cdot v_{ei}^{*}(n) \qquad (12)$$

where

K : constant of adaptation that fixes the algorithm adaptation speed

V_{in} :PA input signal

v_{ei}^{*} : complex conjugate of the error signal described by (3).

Equation (12) can take the equivalent form

$$\frac{a(n) - a(n-1)}{\Delta\tau} = K \cdot v_{in}(n) \cdot v_{ei}^{*}(n) \qquad (13)$$

Taking the limit for $\Delta\tau$ approaching zero and assuming $a(\tau)$ to be analytic,

$$\frac{da}{d\tau} = K \cdot v_{in}(\tau) \cdot v_{ei}^{*}(\tau) \qquad (14)$$

and the actual expression for a is obtained as

$$a(t) = K \int_{o}^{t} v_{in}(\tau) \cdot v_{ei}^{*}(\tau) d\tau \qquad (15)$$

Equation (15) can be simulated easily using DSP based Agilent Ptolemy simulator of Agilent's ADS.

V. SIMULATION RESULTS

Toshiba Microwave Semiconductor's TMD0507-2A Microwave MMIC Amplifier is used for the simulation. Amplifier supplies typically 25 dB gain. The amplifier offers +33dBm P1dB and 22.0dB G1dB. In figure 3, delay characteristics of TMD0507-2A is shown. Delay elements are used in both loops to compensate this delay.

A complex correlater is used to simulate the equation (15) in DSP controller unit. Basic correlater unit has two inputs and outputs. For the first loop inputs are main and error signals to be correlated. And for the second loop inputs are error and feedforward output signals to be correlated. The outputs of correlaters are adaptation coefficients, alpha and beta. These adaptation coefficients applied directly to complex gain adjusters of both loop which is used for adjusting gain and phase to match both upper and lower branches. In figure 4 and 5, adaptation coefficients alpha and beta are shown with respect to time.

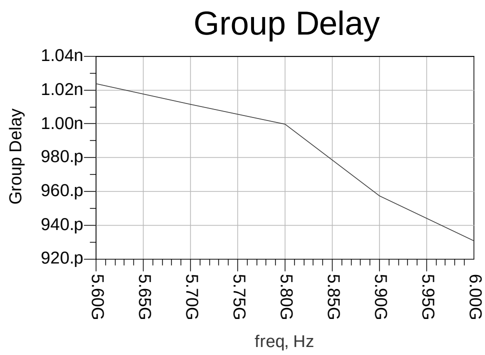

Fig. 3. Delay Characteristic of Used Amplifier

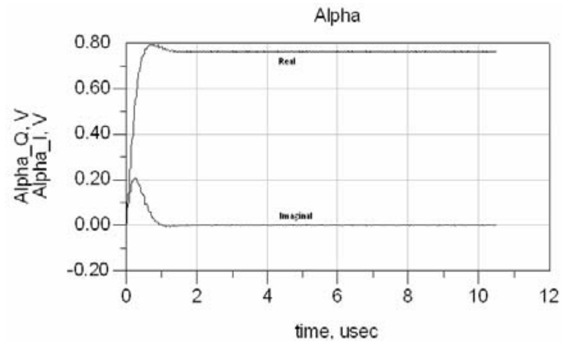

Fig. 4. Adaptation Coefficient Alpha

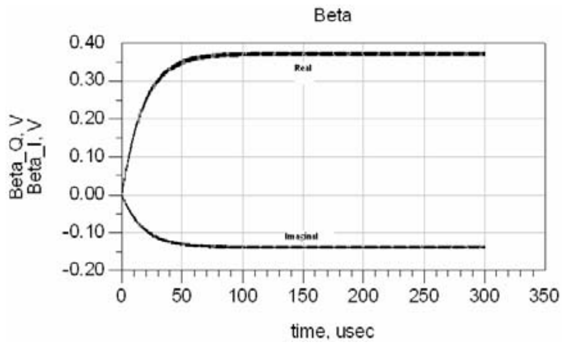

Fig. 5. Adaptation Coefficient Beta

When two-tone signal is applied to input of amplifier, the output is Figure 6. In figure 3rd order IMD is about -30dBc.

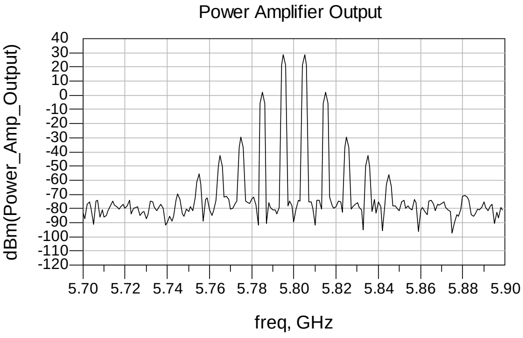

Fig. 6. Power Amplifier Output

In figure 7 and 8, signal cancellation output (error signal) and feedforward output are shown, respectively. It is clearly seen that 3rd order IMD is about -55dBc and 25dB improvement is achieved in IMD performance of amplifier.

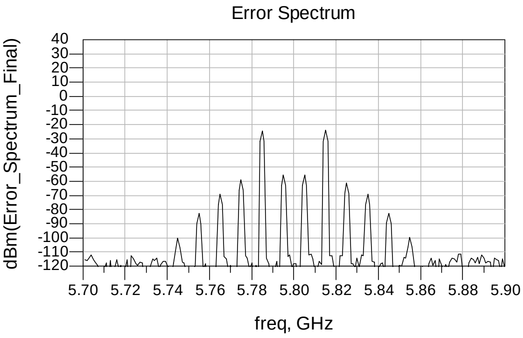

Fig. 7. Error Signal (Signal Cancellation Output)

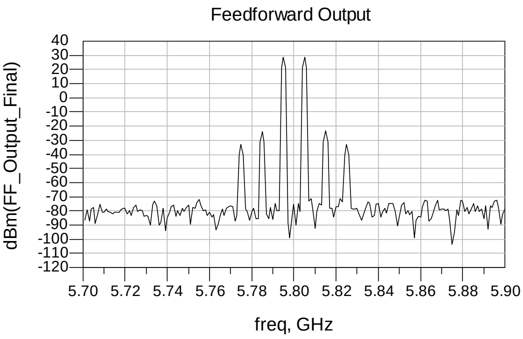

Fig. 8. Feedforward Output

VI. CONCLUSIONS

An adaptive feedforward amplifier with high linearization performance for multi-carrier applications has been designed using simulation made by EDA tools. An acceptable estimate of the complex parameters a and b for the amplitude and phase imbalances compensation in the signal cancellation circuit and the error cancellation circuit, respectively, of a feedforward linearizer was possible, using digital processing of the corresponding signals. A basic adaptive structure based on the LMS algorithm was analyzed and simulated.

REFERENCES

1. James K. Cavers, Adaptation Behavior Of A Feedforward Amplifier Linearizer, IEEE Transactions On Vehicular Technology, Vol. 44, No. 1, February 1995
2. Alfonso J. Zozaya, Eduard Bertran Alberti and Jordi Berenguer-Sau, Adaptive Feedforward Amplifier Linearizer Using Analog Circuitry, Microwave Journal, Vol. 4, pp. 102-114, July 2001
3. Jianyi Zhou, Limin Feng, Xiaowei Zhu nad Wei Hong, Design Of An Ultralinear Wideband Feedforward Amplifier Using EDA Tools, Microwave Journal, Vol. 43, January 2000