Novel DSP Algorithms For Adaptive Feed-forward Power Amplifier Design

Kwok-Po Chan, and Kwok-Keung M. Cheng

Department of Electronic Engineering
The Chinese University of Hong Kong, Shatin, Hong Kong

Abstract — **Much research effort has recently been devoted to the design and implementation of high linearity feed-forward power amplifier for wireless communication systems. This paper presents two novel DSP (digital signal processing) algorithms that can substantially improve the tracking performances of the adaptive control loops. Both computer simulation and experimental results are included for verification.**

I. Introduction

It is well known that feed-forward method provides the highest accuracy and stability for broadband application, in compared to other linearization approaches such as pre-distortion and feedback techniques. The operation of feed-forward system is based on the subtraction of two almost identical signals, and thus this method is very sensitive to the change of operating conditions such as temperature and component ageing. As a result, adaptive control circuitry is often required to compensate for such parameter drift [1].

In the analog implementation of adaptive scheme, the DC offset associated with the electronic components can result in incorrect convergence. To solve this problem, several approaches based on DSP techniques have been proposed recently. The power minimization method [2] is simple to implement, but it suffers from many drawbacks such as long convergence time. Alternatively, the gradient-based adaptation scheme may be employed in which the control parameter is obtained by taking the correlation between the reference and the estimated signals [3]. In [4], the recursive least-square method is also described. In this paper, two new DSP algorithms are proposed to improve the tracking performances of the control loops.

II. Feed-forward Power Amplifier (FFPA)

Fig. 1 shows the basic configuration of a DSP-based adaptive feed-forward power amplifier, which consists of two signal cancellation loops: error extraction loop and error cancellation loop. In the gradient-based adaptive scheme, the complex control parameters are constantly updated by the following equations:

$$\alpha[n]=\alpha[n-1]+K_{\alpha}v_{e}[n]v_{i}^{*}[n] \tag{1}$$

$$\beta[n]=\beta[n-1]+K_{\beta}v_{Lo}[n]v_{e}^{*}[n] \tag{2}$$

where v_i[n], v_e[n] and v_{Lo}[n] are the sampled values of the corresponding down-converted signal waveforms.

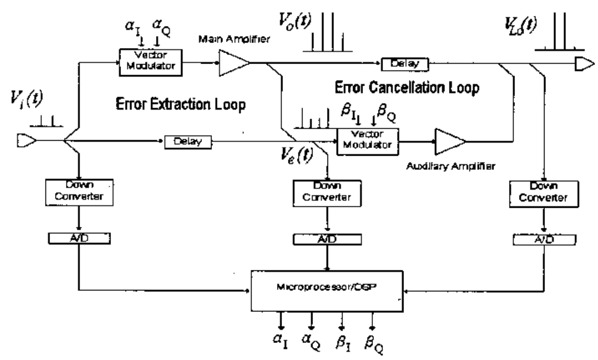


Fig. 1 DSP Based Adaptive FFPA

Furthermore, a main-tone suppression loop [3] may be introduced to reduce the masking effect of the distortion signal by the stronger main tone at the linearized output. This additional loop can be implemented in software as follows:

$$\beta[n]=\beta[n-1]+K_{\beta}v_{Le}[n]v_{e}^{*}[n] \tag{3}$$

$$\gamma[n]=\gamma[n-1]+K_{\gamma}v_{Lo}[n]v_{i}^{*}[n] \tag{4}$$

$$v_{Le}[n]=v_{Lo}[n]-\gamma[n]v_{i}[n] \tag{5}$$

Based on the LMS analysis [5], the value of K_α should be properly chosen to ensure correct convergence:

$$0\le K_{\alpha}\le\frac{2}{P_{i}} \tag{6}$$

Upon convergence, the system can be modeled as a first-order feedback system (Fig. 2) where P_i is the power

0-7803-7695-1/03/$17.00

of the input signal. n_c and α_c are, respectively, the noise estimate and the converged value of $\alpha[n]$ at steady-state. Clearly, the convergence time of $\alpha[n]$ is also governed by the value of K_α. The general tradeoffs between the convergence accuracy and convergence time of the adaptive loop are well described in the literature [1]. Note that similar argument can also be applied to the error cancellation loop and main-tone suppression loop.

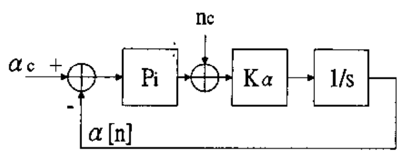


Fig. 2 Error Extraction Loop Model

III. VARIABLE LOOP GAIN ALGORITHM

Here, adaptive loop gain control is introduced for achieving short convergence time and high convergence accuracy simultaneously. In the new algorithm, the updating formulas for the control parameters are modified as follows:

$$\alpha[n]=\begin{cases}\alpha[n-1]+K_\alpha v_e[n]v_i^*[n]\cdot X_\alpha\left|\Delta P_e\right| & \left|\Delta P_e\right|>G_\alpha\\ \alpha[n-1]+K_\alpha v_e[n]v_i^*[n] & \left|\Delta P_e\right|\le G_\alpha\end{cases}\tag{7}$$

$$\beta[n]=\begin{cases}\beta[n-1]+K_\beta v_{Le}[n]v_e^*[n]\cdot X_\beta\left|\Delta P_{Le}\right| & \left|\Delta P_{Le}\right|>G_\beta\\ \beta[n-1]+K_\beta v_{Le}[n]v_e^*[n] & \left|\Delta P_{Le}\right|\le G_\beta\end{cases}\tag{8}$$

$$\gamma[n]=\begin{cases}\gamma[n-1]+K_\gamma v_{Lo}[n]v_i^*[n]\cdot X_\gamma\left|\Delta P_{Lo}\right| & \left|\Delta P_{Lo}\right|>G_\gamma\\ \gamma[n-1]+K_\gamma v_{Lo}[n]v_i^*[n] & \left|\Delta P_{Lo}\right|\le G_\gamma\end{cases}\tag{9}$$

where Δ is the gradient operator; X_α, X_β and X_γ are scaling factors. G_α, G_β and G_γ are threshold constants that determine whether a feedback path with a constant or a variable gain is applied (Fig. 3). Note that a constant loop gain is selected upon convergence and therefore the mean square error (convergence accuracy) is solely determined by the value of K_α.

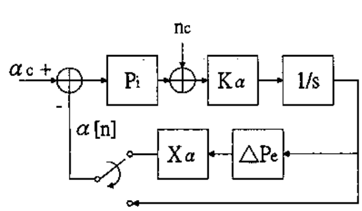


Fig. 3 Error extraction loop model: Variable loop gain

Fig. 4 & 5 show the transient responses of all the loop control parameters obtained by computer simulation (MATLAB program) and experiments (using PHS input signal). Large reduction in the number of iterations required for convergence was observed.

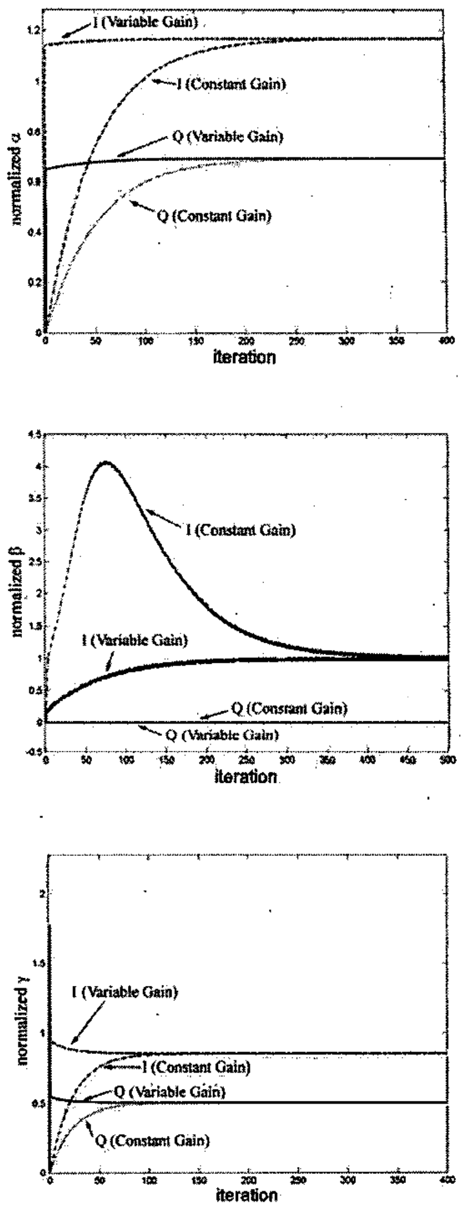


Fig. 4 Simulation results: Variable loop gain

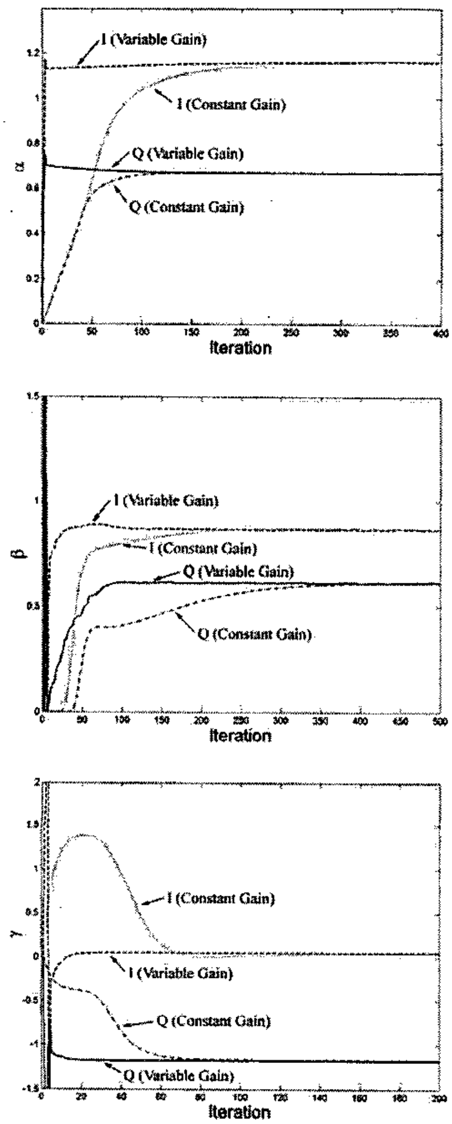


Fig. 5 Experimental results: Variable loop gain

IV. VARIABLE STEP-SIZE ALGORITHM

The second method involves the application of variable step-size determined by the gradient of the control signal itself. Subsequently, the control parameters are updated by the following expressions:

$$\alpha[n]=\begin{cases}\alpha[n-1]+K_\alpha v_e[n]v_i^*[n]+T_\alpha\Delta\alpha & |\Delta\alpha|>S_\alpha\\ \alpha[n-1]+K_\alpha v_e[n]v_i^*[n] & |\Delta\alpha|\le S_\alpha\end{cases}\qquad(10)$$

$$\beta[n]=\begin{cases}\beta[n-1]+K_\beta v_{Le}[n]v_e^*[n]+T_\beta\Delta\beta & |\Delta\beta|>S_\beta\\ \beta[n-1]+K_\beta v_{Le}[n]v_e^*[n] & |\Delta\beta|\le S_\beta\end{cases}\qquad(11)$$

$$\gamma[n]=\begin{cases}\gamma[n-1]+K_\gamma v_{Lo}[n]v_i^*[n]+T_\gamma\Delta\gamma & |\Delta\gamma|>S_\gamma\\ \gamma[n-1]+K_\gamma v_{Lo}[n]v_i^*[n] & |\Delta\gamma|\le S_\gamma\end{cases}\qquad(12)$$

where S_α, S_β and S_γ are the gradient thresholds; T_α, T_β and T_γ are scaling factors. Fig. 6 shows the system model in which an additional feed-forward path is inserted to speed up the convergence rate of the adaptive loop.

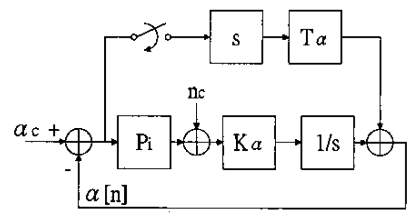


Fig. 6 Error extraction loop model: Variable step-size

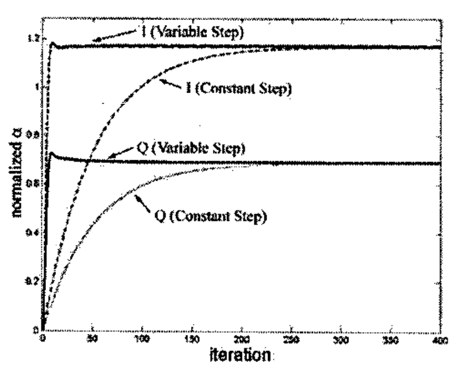
The system transfer function can therefore be derived as:

$$\frac{\alpha}{\alpha_c}=\frac{T_\alpha s^2+P_iK_\alpha}{T_\alpha s^2+s+P_iK_\alpha}\qquad(13)$$

To ensure convergence, the real part of the roots of the denominator should be negative [6], which yields:

$$0<T_\alpha<\frac{1}{4P_iK_\alpha}\qquad(14)$$

Fig. 7 & 8 show the simulated and measured transient responses of the loop control parameters. Reduction in the number of iterations by a factor of 2-10 was found.



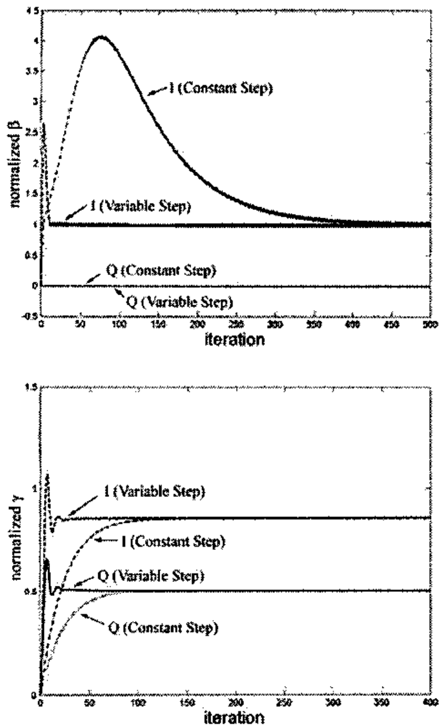
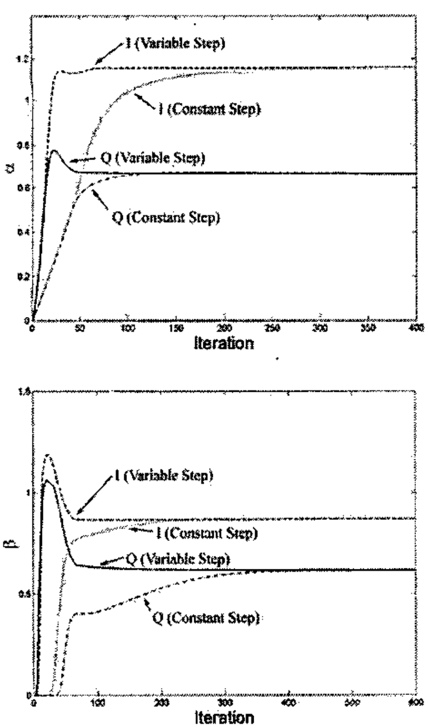


Fig. 7 Simulation results: Variable step-size



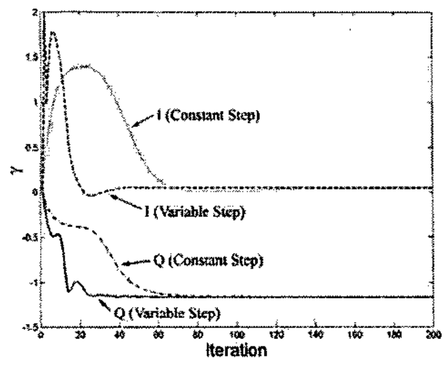


Fig. 8 Experimental results: Variable step-size

V. CONCLUSION

The application of variable loop gain and step-size algorithms to the design of DSP-based feed-forward power amplifier was described. Improvement in convergence speed of loop control parameters was confirmed by computer simulation and measurement results.

REFERENCES

[1] J. K. Cavers, "Adaptive behavior of a feedforward amplifier linearizer", IEEE Transactions on Vehicular Technology, Vol. 44, pp. 31-40, Feb. 1995.

[2] S. J. Grant, "A DSP controlled adaptive feedforward amplifier linearizer", 5th Int. Conf. Universal Personal Communications, Cambridge, MA, vol. 2, pp. 788-792, 1996.

[3] P. Y. Lin, and K. K. M. Cheng "Analysis and simulation of adaptive feedforward power amplifiers for CDMA communication systems", 2001 Asia-Pacific Microwave Conference Proceedings, pp. 988-991, Taipei.

[4] Jiunn-Tsair Chen, "The optimal RLS parameter tracking algorithm for a power amplifier feedforward linearizer, IEEE Transactions on Circuits and Systems-II: Analog and Digital Signal Processing, vol. 46, no. 4, April 1999.

[5] S. Haykin, "Adaptive filter theory", Prentice Hall.

[6] K. Ogata, "Modern control engineering", Prentice Hall.