Pre-distortion Based Linearisation Technique for Power Amplifiers of Wideband Communication Systems

Sangeeta Bawa, Maninder Pal, Jyoti Gupta

Abstract— The high power amplifiers (HPAs) for CDMA and OFDM telecommunication systems need to be highly linear. However in practise, the response of HPA is non-linear which introduces distortion in signals being amplified. This response can be made linear using various linearization techniques. In this paper, an adaptive pre-distortion based linearization technique is introduced. It is based on adding pre-distortions into signals, which will result in cancellation of non-linear distortions appearing in power amplifier. The characteristics of this non-linear adaptive linearizer are represented by an analogue polynomial. In this paper, the coefficients of this polynomial are adjusted automatically in a manner that the output of linearizer when fed to an amplifier, the final output becomes error free and the whole system behaves like a linear amplifier. This helps reducing the distortion in power amplifiers, resulting in a better linearised output.

Index Terms— Pre-distortion, High power amplifier, Linearization, Non-linear amplifier, Adaptive Linearizer, Perturbation based method.

1 Introduction

RECENTLY, modern communication services such as OFDM and WCDMA have created a huge demand for highly linear high power amplifiers [1-2]. It is primarily because these services require high power amplifiers supporting wideband communications. In addition, these services also involve the transmission of multiple signals and/or large quantities of information at high data rates. In HPAs, distortion of signals is usually produced by its non-linear response. This distortion is usually measured in terms of amplitude, although it also includes associated abnormalities in phase [3]. For phase, the associated distortion is called phase non-linearity. The term *amplitude non-linearity* can be considered as a measure of deviation from the desired straight-line behaviour and can be represented by the following power series:

$$V_{out}= K_1V_{in}+ K_2V_{in}^2 + K_3V_{in}^3....+ K_nV_{in}^n \quad (1)$$

Where, $K_1, K_2, K_3,...,K_n$ are the coefficients of non-linear polynomial, and V_{in} and V_{out} are the respective input and output of HPA. In situation, when a single carrier input signal, (say, a sine wave), is substituted into this expression (1), the output waveform will contain the original sine wave and its harmonic distortion products [4]. The harmonics can be eliminated by filtering and do not pose a problem except for wideband communication applications requiring wide bandwidth. However, when more than one carrier is present, additional new signals known intermodulation distortions (IMDs) are produced in the vicinity of input signals. The IMDs are located at frequencies above and below the input carriers, and at frequency intervals equal to the separations of input carriers. Filtering cannot easily eliminate IMD products, as these are located on the same frequency or near to the desired input signals. Therefore, a wide frequency range of operation of linearizer and high power amplifier is highly desirable.

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2 Linearization Techniques

Linearization can be referred as a procedure for reducing signal distortions produced by the non-linear response of an amplifier. Several engineers and researchers have proposed and designed many models of linearizers such as feed-forward [3], feed backward [5], envelope feedback system [6], cartesian feedback system [7] and predistortion [7-9]. However, these are associated with several drawbacks. The most common of these drawbacks include: phase delay, reduction of the achievable bandwidth, degradation of gain and phase margin, and low power efficiency for low input signals [10]. Among the above mentioned linearization techniques, pre-distortion is the most commonly used. It is based on the concept of inserting a nonlinear module between the input signal and power amplifier [11-12]. The nonlinear module generates IMD products that are in anti-phase with the IMD products produced by the power amplifier. These anti-phase IMD signals ideally cancels the IMDs produced by HPA; and thus, resulting in a better overall linearised behavior of the whole system. In this paper, an adaptive linearizer is implemented using perturbation based method; whereby, the characteristics of linearizer are represented by non-linear polynomials.

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3 Adaptive Linearizer

The principle of an amplifier with an adaptive linearizer is explained in Fig. 1. The output of non-linear amplifier is fed back to the error detector after attenuating through a fixed attenuator. The error, due to non-linearity of amplifier, is calculated by taking the difference of the attenuated output of the non-linear amplifier and input signal. This error is fed to the adaptive linearizer element, which varies the envelope of input signal (V_{in}) resulting in the signal (V_{dout}). This V_{dout} signal is fed to the amplifier such that error(t) can be minimized. In this adaptive linearization technique, perturbation based method is used to adjust the coefficients of polynomials in such a way that the error approaches to zero and the output of an overall system (amplifier and linearizer) becomes closely to linear.

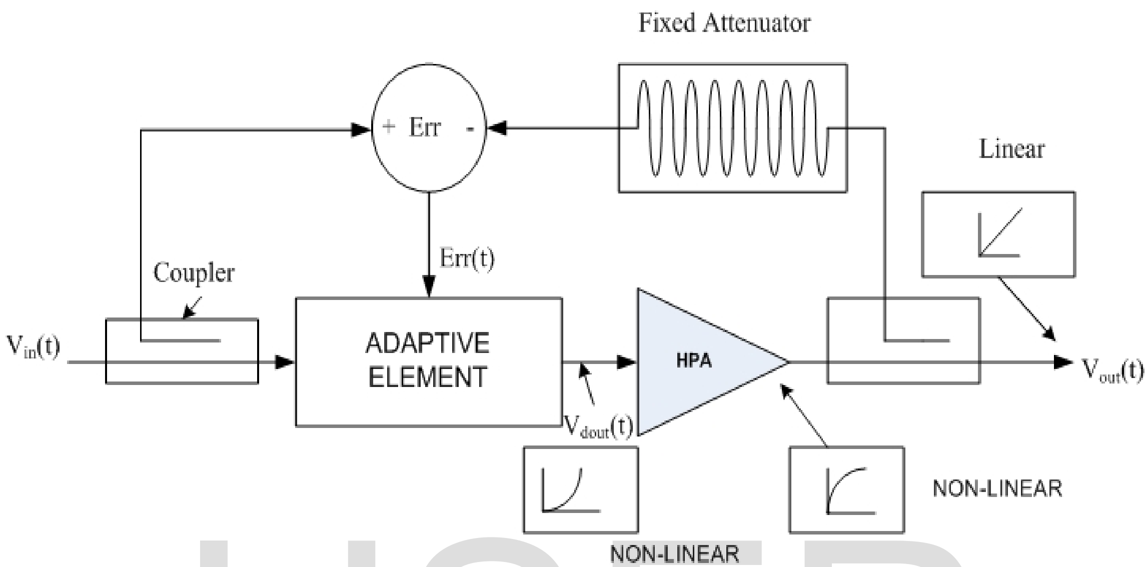

Fig. 1. Block diagram of a high power amplifier using the perturbation based adaptive linearization technique.

The transfer characteristics of a HPA and adaptive linearizer shown above in Fig. 1 can be expressed as polynomial. In general, for any input x, the polynomial of n^{th} order is given by:

$$y(x) = w_0 + w_1 x + w_2 x^2 + w_3 x^3 + ... + w_{N-1} x^{N-1} + w_N x^N = \sum_{i=0}^{N} w_i x^i \quad (2)$$

where, $w_0, w_1, w_2, ..., w_N$ are the coefficients of polynomial. According to the pre-distortion linearization theory, the response of linearizer is made equal in amplitude and antiphase of the IMDs generated by the amplifier [9]. As shown in Fig. 1, the characteristics of HPA represents closely to tangent hyperbolic functions; which can be represented in polynomial form (say, up to 5^{th} order) as:

$$\text{Tanh}(t) = 0.9971*t - 0.3076*t^3 + 0.0725*t^5 \quad (3)$$

It can be interpreted from (3) that it comprises only of odd terms and the coefficients of each of the alternate terms after the third order term change the sign alternatively. Therefore, the general expression for characteristics (as shown in Fig. 1) of HPA can be represented as:

$$V_{out} = w_0 + w_1(V_{dout}) + w_3(V_{dout})^3 + w_5(V_{dout})^5 + ... + w_{2*N-1}(V_{dout})^{2*N-1}$$
$$= w_0 + \sum_{i=0}^{N} w_{(2*i-1)}(V_{dout})^{(2*i-1)} \quad (4)$$

where, V_{dout} and V_{out} are respectively the input and output of the non-linear amplifier shown in Fig. 1. As the characteristics of linearizer in Fig. 1 is anti-phase to nullify the effect of higher order-terms in (expression 4); therefore, the polynomial representing the characteristics of linearizer can be expressed as:

$$V_{dout} = a_0 + a_1 V_{in} + a_3(V_{in})^3 + a_5(V_{in})^5 + ... + a_{2*N-1}(V_{in})^{2*N-1} \quad (5)$$

The coefficients of this polynomial (expression 5), representing the characteristics of adaptive linearizer are adjusted in such a way that the characteristics of adaptive linearizer are inverse to that of the non-linear amplifier. If the output signal from such a linearizer is fed to the non-linear amplifier; then it will give a linearised output. The value of the coefficients a_0, a_1, a_3,... of the adaptive linearizer are calculated using the algorithm based on perturbation based approximation method, mentioned below and as shown in Fig. 2.

4 Algorithm

1. Start if there is an error i.e. error~0. Calculate the Error for one complete cycle and take this as the initial error (Error_initial). The cycle time depends upon the input signal time period.
2. Increment the first coefficient of adaptive linearizer polynomial, a_1 by Δ i.e. $a_1 \rightarrow a_1 + \Delta$, where, Δ is the step size (Initially, a_1 =1 and all other coefficients are zero).
3. Calculate the error (Error′) per cycle of the input waveform. Compare this error with the initial error (Error′_initial).
4. If this error (Error′) is less than the initial value (Error_initial), i.e. initial error, then consider this error as the reference error (Error_Reference) and value of a_1 remains the same (as in step 2) else if the Error′ is increased, then the steps 2 to 4 is repeated by decrementing the value of a_1 by 2Δ, i.e. $a_1 \rightarrow a_1$-2Δ.

5. If, say, again the error (Error′) comes out to be more than the initial error (Error_initial) the value of the coefficient a_1 is changed to its original value. It is so because the error is not reduced by both incrementing and decrementing the value of a_1 by Δ.

6. The process is repeated for the next coefficient a_3. This algorithm compares the new error (Error′), for change in coefficient a_3, with the reference error (Error_Reference). If this error (Error′) is less than both the initial error (Error_initial) and the reference error (Error_Reference), then this error (Error′) is considered as the reference error (Error_Reference) for the next step. The value of a_3 remains the same, but if the Error′ is increased, then the algorithm compares the error by decrementing the value of a_3 i.e. $a_3 \rightarrow a_3$-2Δ.

7. If the error is not reduced to zero even by changing the Nth coefficient; then the whole process, i.e. from steps 2 to 6, is repeated by decreasing the value of Δ in order to Δ/2, Δ/4, Δ/8 and so on till the error becomes zero. The value of Δ is changed in a particular fashion shown in Fig. 3.

8. This process is iterative and repeats itself till the coefficients a_0, a_1, a_3 etc are changed to such a value so that the error (Error′) becomes zero i.e. the output of the amplifier becomes linear. The flowchart for this method is shown in Fig. 3.

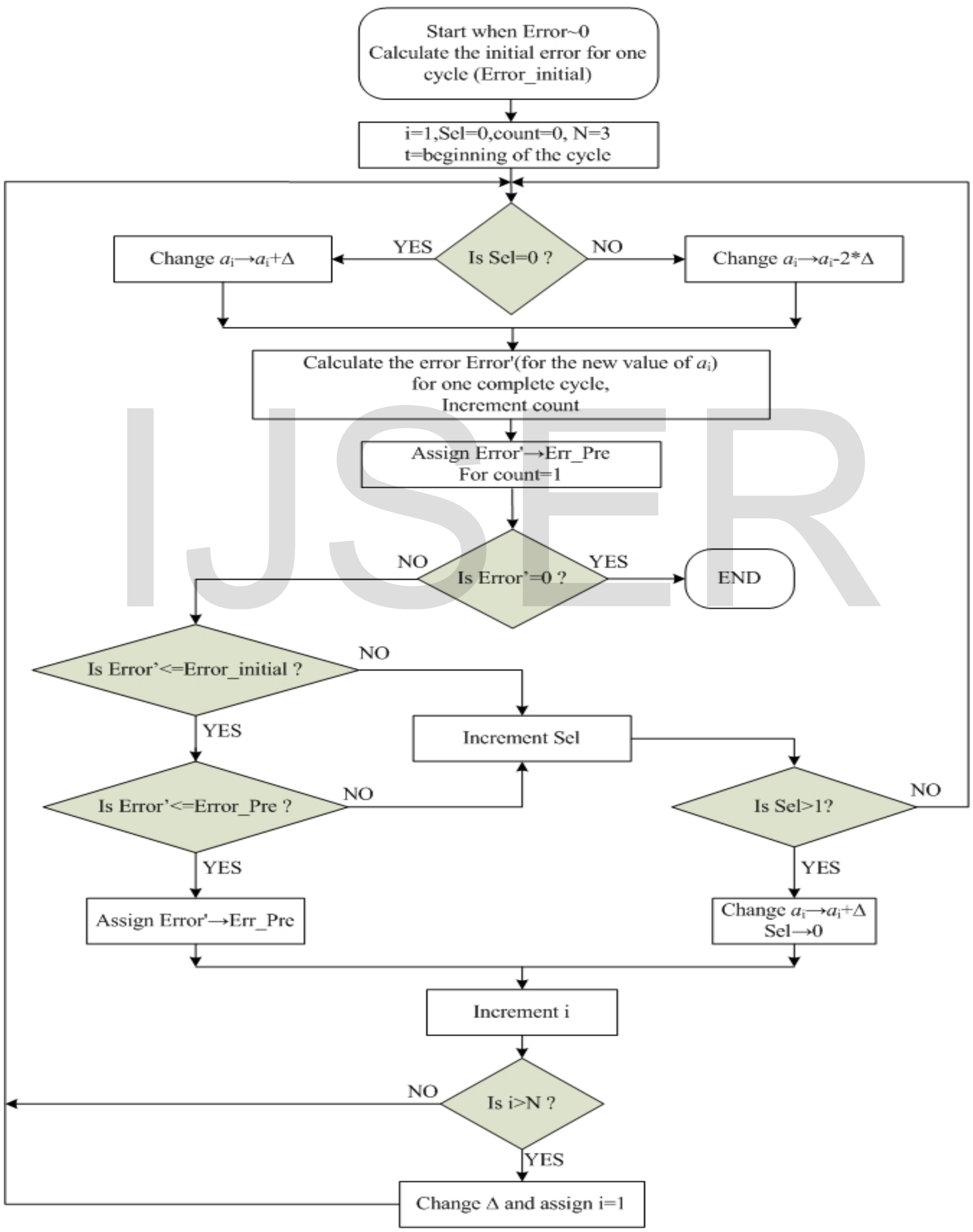

Fig. 2. Flowchart for the perturbation based approximation method for adaptive linearizer.

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The value of step size delta is changed after every complete execution of the program i.e. after changing the value of all the coefficients. The initial value of delta can be calculated mathematically by assuming that the polynomial for the amplifier is non-linear tangent hyperbolic function i.e.

$$V_{out} = g(f(V_{dout})) = \tanh(V_{dout}) = 0.9971*(V_{dout}) - 0.3076*(V_{dout})^3 + 0.0725*(V_{dout})^5 \quad (6)$$

From expression (6) $w_5 = 0.0725, w_3 = 0.3076, w_1 = 0.9971$ and $w_3 / w_1 = 0.3076/0.9971 = 0.3085$. This means the error is:

$$V_{error} = (a_3 - w_3/w_1)V_{in}^3 - (3a_3w_3/w_1)V_{in}^5 - (3a_3^2w_3/w_1)V_{in}^7 - a_3^3w_3/w_1V_{in}^9 \quad (7)$$

Substituting $a_3 = \Delta$, for first execution of the algorithm to have

$$V_{error} = (\Delta - 0.3085)V_{in}^3 - (0.9255*\Delta)V_{in}^5 - (0.9255*\Delta^2)V_{in}^7 - (0.3085*\Delta^3)V_{in}^9 \quad (8)$$

In expression (8), the third order term causes the bending of transfer characteristics of amplifier; while, the higher order terms causes the characteristics to bend near the saturation. So, the major contribution to the nonlinearity is due to the third order term. In expression (8), as per the perturbation theory, the major contribution to the error is due to the third order term, so the value of delta should be such that after first execution of the program the first order term becomes zero i.e.

$$\Delta - 0.3085 = 0 \quad \rightarrow \quad \Delta = 0.3085$$

This makes,

$$V_{error} = -(0.2855)V_{in}^5 - (0.0880)V_{in}^7 - (0.0090)V_{in}^9 \quad (9)$$

At each stage, the error is compared with the initial error and with the previous error. For example, let the coefficient *a*3 is changed when the delta changes from Δ-Δ/2+Δ/4 to Δ-Δ/2+Δ/4-Δ/8 or Δ-Δ/2+Δ/4+Δ/8. The algorithm will then compare the error (Error (Δ-Δ/2+Δ/4-Δ/8)) with the initial error and also with previous error (Error (Δ-Δ/2+Δ/4)). If the error due to step size Δ-Δ/2+Δ/4-Δ/8 is less than the initial error and error (Error (Δ-Δ/2+Δ/4)) i.e. previous error, then the algorithm will change the value of the coefficient *a*3, else the algorithm will repeat for Δ-Δ/2+Δ/4+Δ/8. If in both cases, the error is more, then the value of coefficient is not changed and the algorithm will go for the next coefficient. The algorithm will only change the coefficient, if it finds a decrease in error than the previous error value. The procedure for changing step size Δ is shown in Fig. 3 and Fig. 4.

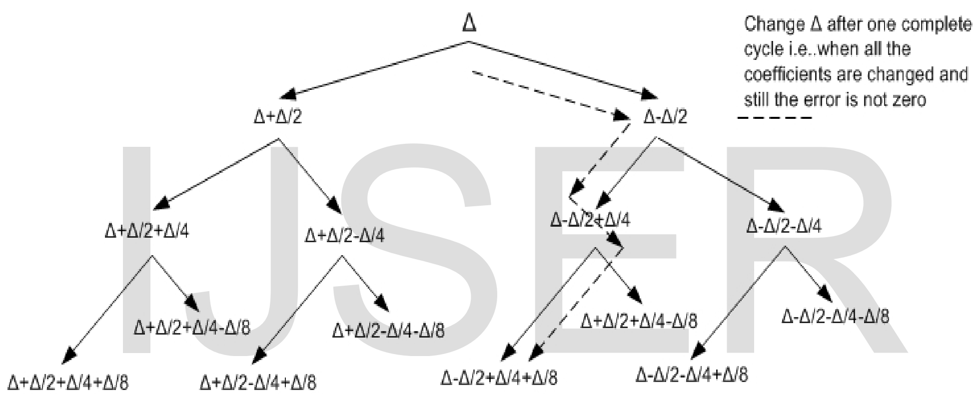


Fig. 3. Step size delta.

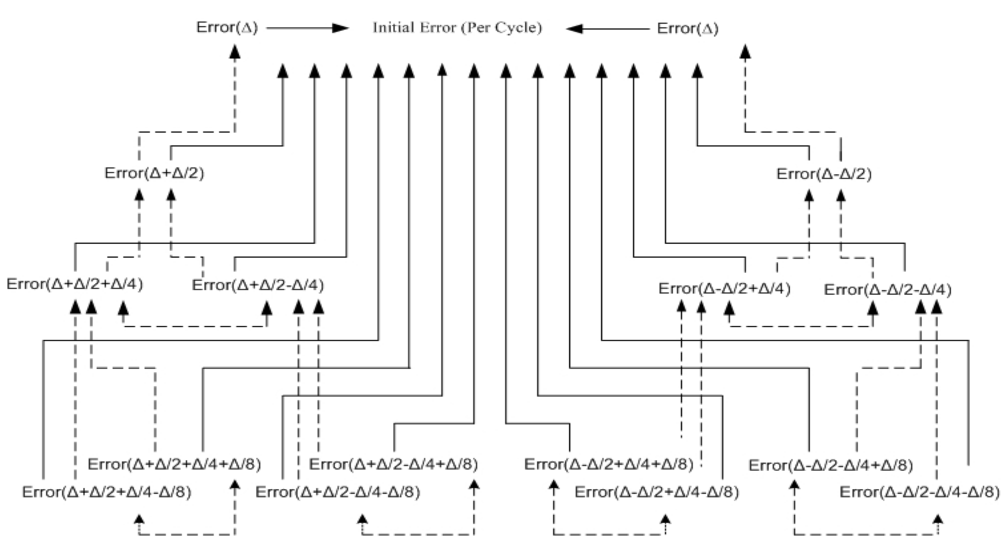
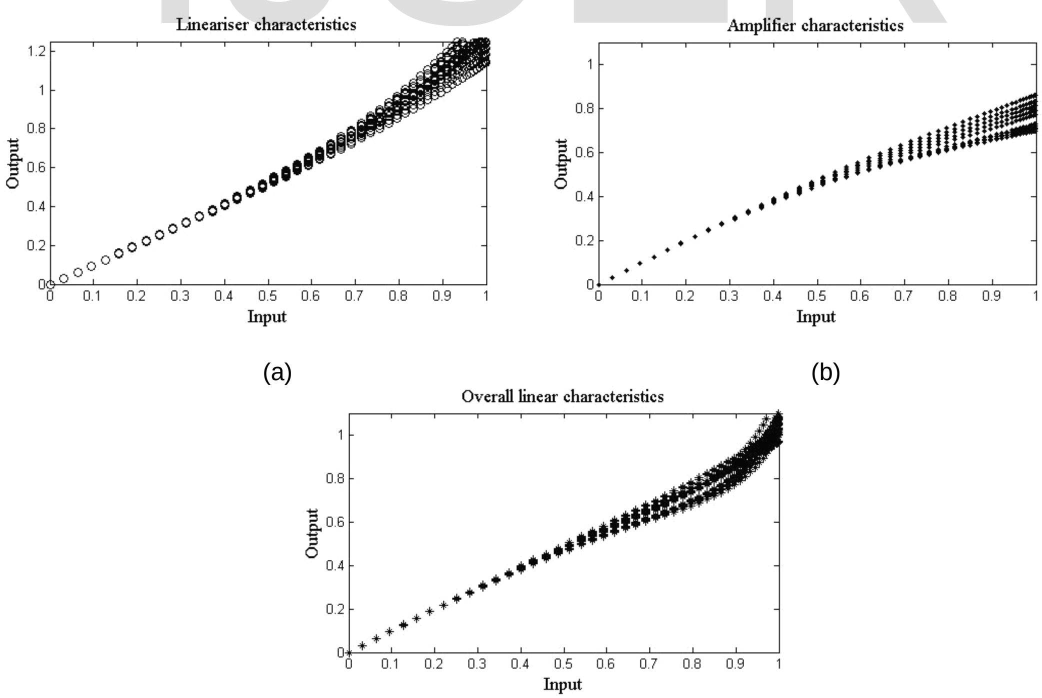


Fig. 4. Error per step size delta.

5 RESULTS & DISCUSSIONS

To determine the performance of the proposed perturbation based adaptive linearizer, extensive simulations have been performed in MATLAB. The proposed algorithm is evaluated using many different signals such as BPSK, QPSK and QAM. However in this paper, a simulated QAM signal at 1000 MHz is used. For simplicity, the non-linear AM-AM distortion function of HPA is taken up to only fifth-order polynomial. It is also because higher order terms have negligible effects so these are neglected. Fig. 5(a) shows the simulated non-linear amplifier characteristics for different values of coefficients (w_1, w_3 and w_5) of fifth-order polynomial. The respective characteristics of linearizer and the overall system are shown in Fig. 5(b) and 5(c). It is to be noted that these characteristics of the simulated non-linear amplifier can be changed by using different values of coefficients. The characteristics of linearizer can be changed by using different step size value (Δ) and number of iterations used for adaptation. From the extensive simulations performed, it is found that for the initial value of step size greater than w_1 / w_3, the percentage of error is higher in the beginning compared to the value of step size less than w_1 / w_3. It is also found that if the initial value of step size is large in comparison to value of w_1 / w_3, the algorithm will take a comparatively large number of cycles to minimize the error.

It can also be observed from Fig. 6, that the linearizer characteristics are opposite to the amplifier characteristics, in order to nullify its non-linear effect. For simplicity, the gain of amplifier is set to unity. The non-linear transfer characteristics of HPA showing the effect of non-linearity especially near compression can be seen in Fig. 5 and 6; where, the output power reaches saturation when the input exceeds a certain power level. From Fig. 5 and 6, it is evident that the algorithm successfully gives a better linearised output up to the saturation point of PA characteristic. To investigate this, a 64-M QAM signal (shown in Fig. 7) is fed to the designed amplifier system (amplifier and linearizer) for amplification. The signal is first fed to linearizer in which the perturbation based algorithm gives the locus of the values of step size and the coefficients of polynomial of linearizer for which error is less than the initial error. Figures 7 - 14 demonstrates the effects of PA nonlinearity on the signal constellation, linearizer output and the final signal constellation realigned with the input constellation. It can be seen from Fig. 7 to Fig. 10 that the non-linearity of amplifier affects primarily on the outer signals in the constellation plot. It is because these are closer to the compression region in the transfer characteristics of HPA (Fig. 6). The linearizer has transfer characteristics in anti-phase to those of amplifier, so these non-linear characteristics displaced the signals towards centre of constellation, as shown in Fig. 9. When this signal is fed to the amplifier, the already shifted signal by linearizer nullify the shift due to the non-linearity of the amplifier transfer characteristics, which finally results in a better linearised output (shown in Fig. 10). Consider another example in Fig. 11, in which noise is added to the above simulated QAM signal. The energy-per-bit (E_b) per noise power spectral (N_0) density was 35dB. This demonstrates that the predistorter is able to reduce the EVM by about a factor of three even when transmitting a noisy signal. The signals at various stages (amplifier, linearizer and overall output) are shown with and without predistortion, in Figures 11 to 14. It can be evident from Figures 11 to 14 that the proposed algorithm works well even in the presence of a high level of noise.



(a) (b)

(c)

Fig. 5. (a) Linearizer, (b) non-linear amplifier, and (c) the predistorter linearizer based amplifier's AM-AM characteristics.

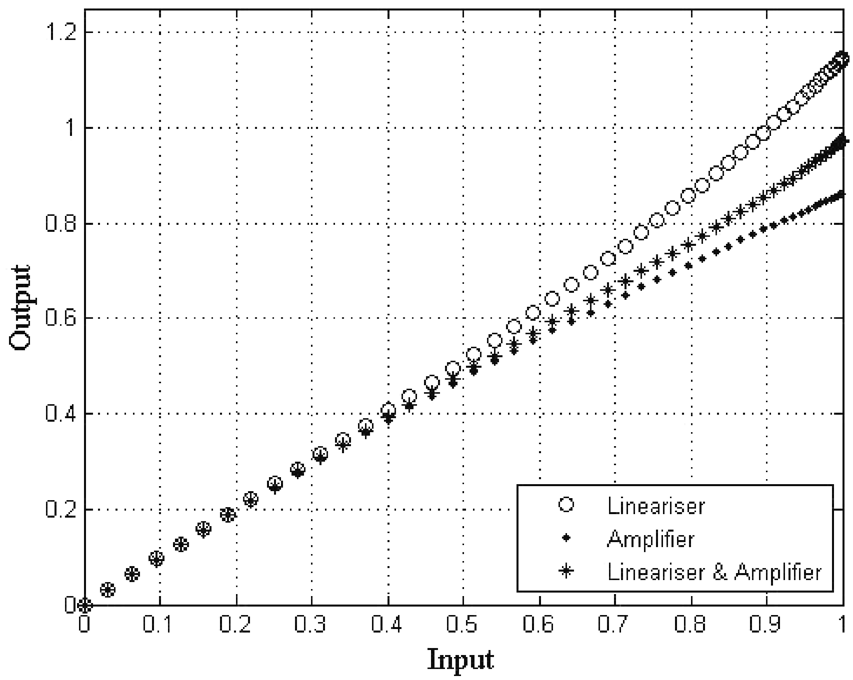


Fig. 6. A typical example of the characteristics of a non-linear amplifier, corresponding linearizer and the overall response.

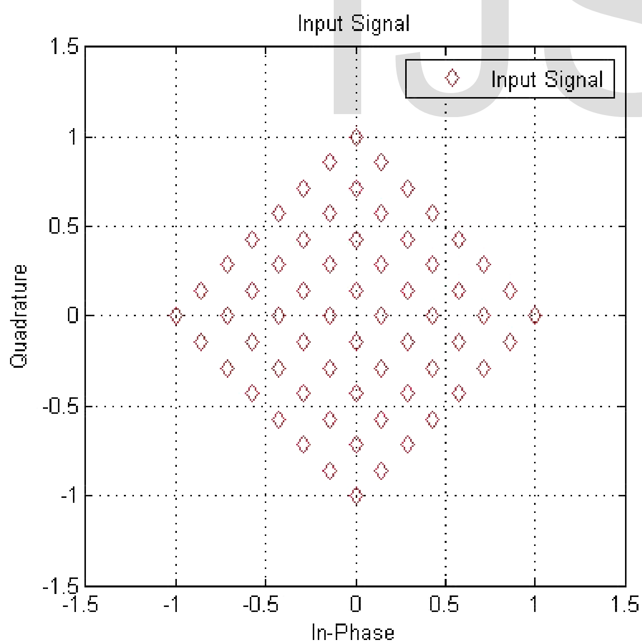


Fig. 7. The input 64-M QAM signal.

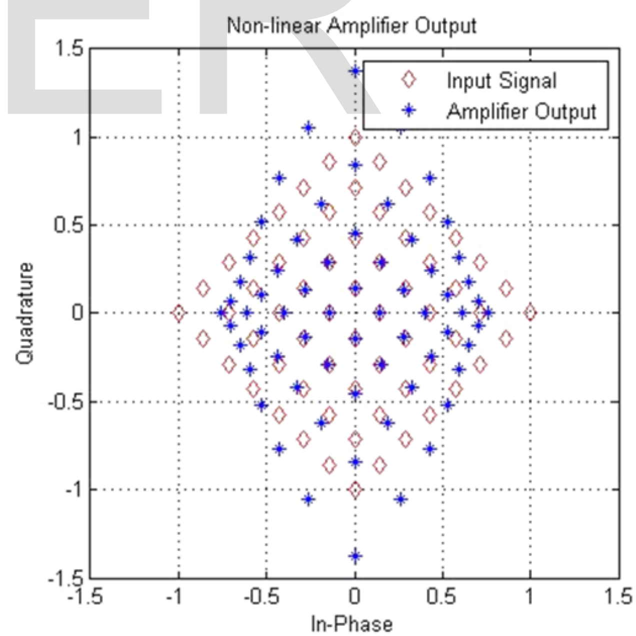


Fig. 8. The input 64-M QAM signal and the scaled non-linear amplifier output showing the effect of gain compression on outer symbols.

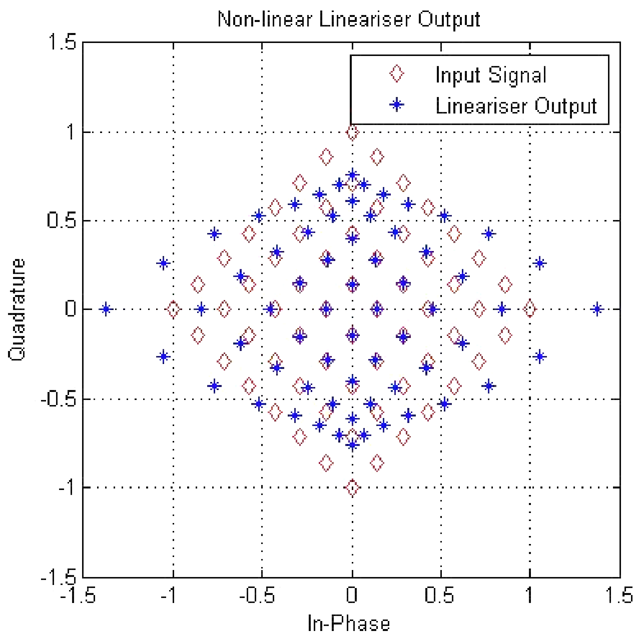


Fig. 9. The input 64-M QAM signal and the scaled non-linear linearizer output affecting the outer symbols.

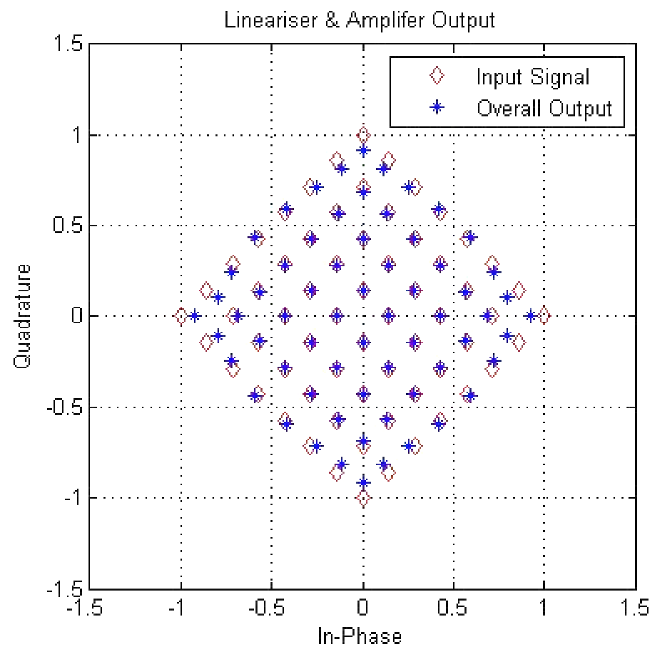


Fig. 10. The input 64-M QAM signal and the scaled pre-distortion based linearizer and amplifier output.

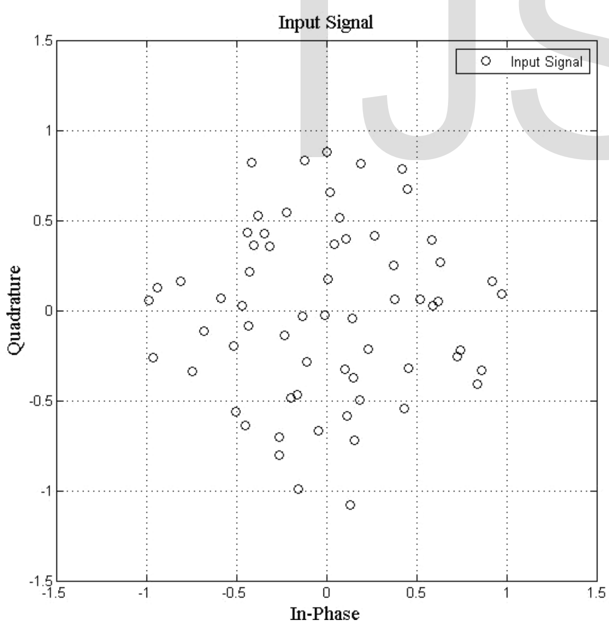


Fig. 11. The input 64-M QAM signal with noise.

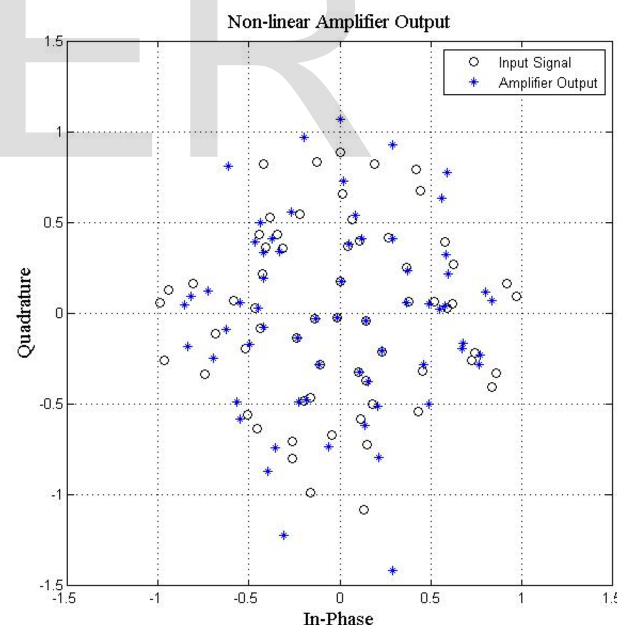


Fig. 12. The input 64-M QAM signal and the scaled non-linear amplifier output showing the effect of gain compression on outer symbols.

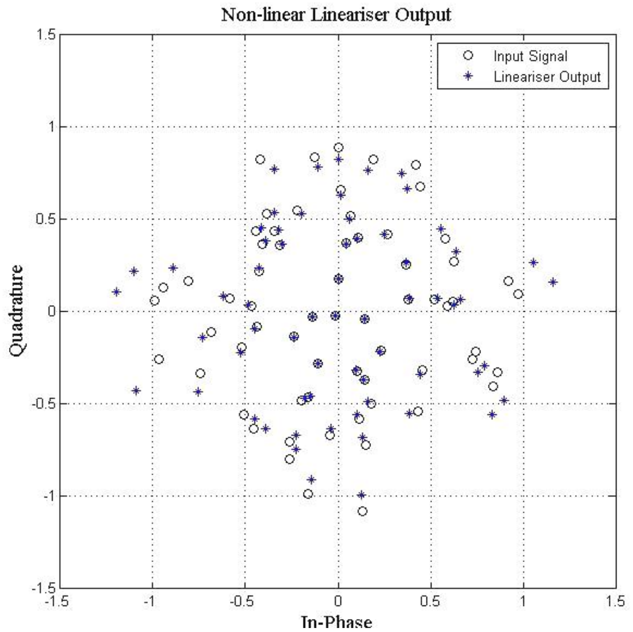


Fig. 13. The input 64-M QAM signal and the scaled non-linear linearizer output showing the effect on outer symbols.

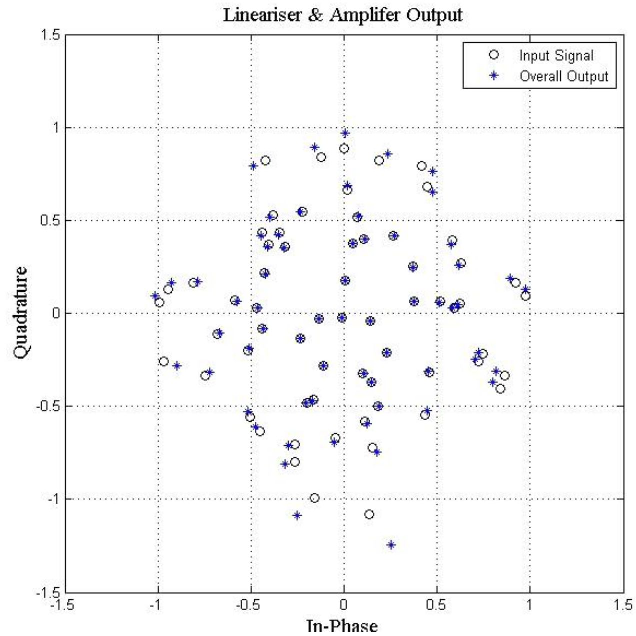


Fig. 14. The input 64-M QAM signal and the scaled predistortion based linearizer and amplifier output.

6 Conclusion

In this paper, a pre-distortion based adaptive linearization technique is proposed. In this technique, the transfer characteristics of linearizer are represented by a polynomial and its coefficients are adjusted by using a perturbation based approximation method. It is found from the analytical studies that the linearizer polynomial consists of only odd order terms. The coefficients of this polynomial are adjusted in a manner that the characteristics of linearizer become opposite to that of amplifier. This will cancel the effect of non-linear response of amplifier, and gives a better linearised output. The error at each instant is compared with the initial error and with the previous error. The output of this system is nearly linear but if the degree of nonlinearity is increased, then the algorithm takes a comparatively large number of cycles to reduce distortion near the saturation. It has been noted that the linearizer works effectively and fast for low degree of distortion and even in presence of noise. From the extensive simulations performed, the initial value of step size suggested to choose is between 0.3 and 0.5. Thus from the results, it can be concluded that the proposed pre-distortion approach has potential in improving the performance of amplifiers and is able to linearize the amplifier up to points close to saturation.

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