PAPER

Adaptive Digital Predistortion with Iterative Noise Cancelation for Power Amplifier Linearization*

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SUMMARY In this paper, we propose a power amplifier linearization technique combined with iterative noise cancelation. This method alleviates the effect of added noises which prevents the predistorter (PD) from estimating the exact characteristics of the power amplifier (PA). To iteratively cancel the noise added in the feedback signal, the output signal of the power amplifier without noise is reconstructed by applying the inverse characteristics of the PD to the predistorted signals. The noise can be revealed by subtracting the reconstructed signals from the feedback signals. Simulation results based on the mean-square error (MSE) and power spectral density (PSD) criteria are presented to evaluate PD performance. The results show that the iterative noise cancelation significantly enhances the MSE performance, which leads to an improvement of the out-of-band power suppression. The performance of the proposed technique is verified by computer simulation and hardware test results.

key words: *adaptive digital predistorter, iterative noise cancelation, linearization techniques, inverse look-up table*

1. Introduction

The predistortion (PD) technique has become an effective linearization technique [1], [2] which could exploit fully the power of the amplifier while maintaining a good linearity characteristic. The typical adaptive PD utilizes feedback signal to estimate the nonlinearity of the high power amplifier (HPA). Owing to the high adaptability based on DSP implementation, the digital predistorter (DPD) is considered as one of the most promising method for implementing linearization technique.

The actual DPD can be mainly accomplished by using polynomial based approach [3], [4] or look-up tables (LUT) based approach [5], [6]. The DPD creates a version of the desired modulation making use of the feedback signals of the actual HPA output, and the inverse of the HPA characteristics for the DPD to compensate for the AM/AM and AM/PM characteristics of the amplifier is composed into the polynomial coefficients or LUT.

Despite many studies on the predistortion techniques, there has been no in-depth study on the noise effect. The main contribution of our work is to provide the solution of predistortions in the realistic noisy environment. It deserves special emphasis that the iterative noise cancelation can be combined with any legacy method, by simply adding the iterative noise canceling module. In this paper, we propose an effective power amplifier linearization technique combined with an iterative noise cancelation. This method alleviates the effect of additional noises which prevents DPD from estimating the exact characteristics of HPA. To iteratively cancel the noise added in the feedback signal, the output signal of power amplifier without noise is reconstructed by applying the inverse characteristics of PD to the predistorted signals. The noise can be revealed by subtracting the reconstructed signals from the feedback signals. Simulation results based on the mean-square error (MSE) and power spectral density (PSD) criteria are presented to evaluate PD performance. The results show that the iterative noise cancelation significantly enhances the MSE performance, which leads to an improvement of the out-of-band power suppression.

2. Adaptive DPD with Iterative Noise Cancelation

2.1 Legacy Adaptive Predistortion Technique

Figure 1 illustrates the block diagram of the adaptive DPD and HPA [10]. The predistorter aims to manipulate transmit signals prior to amplification by the power amplifier, which inverts the distortion characteristics the power amplifier. If the nonlinear property of the power amplifier $\mathbb{G}$ is corrected by combining the predistorter $\mathbb{F}$, the overall system can be described by the ideal PA as $z(n) = \mathbb{G}(\mathbb{F}(x(n))) = kx(n)$,

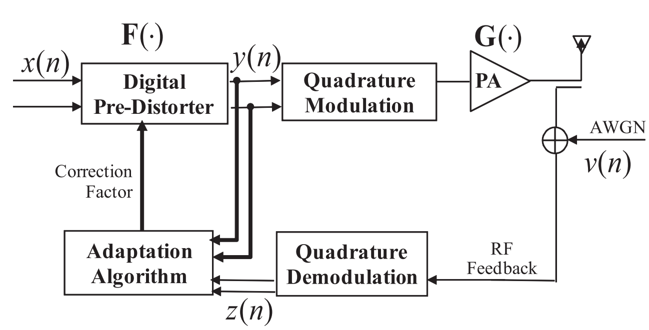

Fig. 1 Block diagram of the generic digital predistortion.

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where k is a pre-specified linear amplification constant.

Correction coefficients stored in DPD are commonly calculated by comparing the predistorted (reference) signal sampled at the output of DPD and RF feedback signal sampled at output of a transmitter. Both signals are feedback to an adaptation block as depicted in Fig. 1. Adaptive correction is realized by using real-time transmit signal flow for the coefficient calculation. In the feedback loop, the output of the PA is downconverted and compared with the delayed version of the predistorted signals as reference. To estimate the delay in the RF feedback loop, correlation between the reference signal and the feedback signals is performed, and the delay is imposed to the reference signal.

The correction factors are computed based on the adaptation algorithm, and they are iteratively updated to reduce errors between the HPA model $\mathbb{F}^{-1}(\cdot)$ and the actual HPA $\mathbb{G}(\cdot)$. Since the DPD coefficients are obtained by measuring the difference between the predistorted signal $\mathbb{F}(x(n))$ and the output of the actual HPA $\mathbb{G}(\mathbb{F}(x(n)))$, estimating the HPA characteristics $\mathbb{G}(\cdot)$ cannot be affected even though coefficients are updated into $\overline{\mathbb{F}}(\cdot)$ during the period. Therefore, at each iteration, adaptation on the function of DPD $\mathbb{F}(\cdot)$ and calculating the error can be carried out simultaneously.

2.2 Effect of the Additive Noise

Example 1: Suppose the initial DPD $\mathbb{F}(x) = ax$ and the actual HPA $\mathbb{G}(x) = bx$ for input symbol x. The adaptive algorithm updates $\mathbb{F}(x)$ with a constant step-size $\mu = 1$. The noise constantly affects as an amount of v. An explicit expression for the estimated HPA can be given as follows.

Solution: The alteration of the predistorted signal $y(n)$, the feedback HPA output $z(n)$ and DPD coefficient $\mathbb{F}(n)$ according to each itaration n can be listed as

n	1	2	$\cdots$	n
$y(n)$	ax	$\mathbb{F}(1)x$	$\cdots$	$\mathbb{F}(n-1)x$
$z(n)$	$bax+v$	$b\mathbb{F}(1)x+v$	$\cdots$	$b\mathbb{F}(n-1)x+v$
$\mathbb{F}(n)$	$\frac{1}{b+\frac{v}{ax}}$	$\frac{1}{b+\frac{v}{\mathbb{F}(1)x}}$	$\cdots$	$\frac{1}{b+\frac{v}{\mathbb{F}(n-1)x}}$

If the adaptive algorithm is infinitely updated, i.e., n is sufficiently large, $\mathbb{F}(n)$ is asymptotically equal to $\mathbb{F}(n-1)$, $\lim_{n\to\infty}\mathbb{F}(n) = \lim_{n\to\infty}\mathbb{F}(n-1)$ so that $\mathbb{F}(n)$ can be re-written as

$$\frac{1}{t} = \frac{1}{b + \frac{1}{t\mathrm{SNR}}} \;\Rightarrow\; t^2 - bt - \frac{1}{\mathrm{SNR}} = 0 \tag{1}$$

where $t = b + \frac{v}{\mathbb{F}(n-1)x}$ and $\mathrm{SNR} = x/v$. Consequently, from quadratic formula, the estimated HPA t, the root of quadratic equation, can be obtained as

$$t = \frac{b}{2} + \frac{1}{2}\sqrt{b^2 + \frac{4}{\mathrm{SNR}}}. \tag{2}$$

Although the actual and expected HPA is b, the exact property of HPA cannot be obtained. This explains how the DPD performance is degraded due to the noise effect, which leads to higher spectral growth in out-of-band. ■

As discussed in *Example 1*, the noise effect should be overcome to alleviate the nonlinear distortion caused by HPA. As a solution, in this paper, iterative noise cancelation technique is proposed. By reducing the noise in (2), the actual HPA characteristics can be obtained as $t \to b$.

2.3 Proposed Technique for Iterative Noise Cancelation

The block diagram of proposed DPD combined with an iterative noise cancelation is shown in Fig. 2. The noise canceling module, the dotted rectangle, is added on the conventional DPD. That is, the proposed method is simple to implement and applicable to any conventional scheme to improve the performance.

The key to the DPD with iterative noise cancelation lies in the reduction of the additive noise. The noise can be detected by estimating the output signal of HPA. The signal generation is shown in the noise canceling module of Fig. 2. Using the inverse LUT, the estimated output signal of HPA is generated and compared with the actual output signal of HPA $z(n)$. The noise is estimated by canceling the estimated output signal of HPA from the actual output signal of HPA. After an initial estimation of a noise with the help of the estimated output signal of HPA generated by using the known transmitted signal, the noise is iteratively subtracted and estimated to enhance the reliability of estimation, updating the DPD block. Therefore, we will have the exact information of noise for every iteration.

2.4 Pseudo-Code for Iterative Noise Cancelation Algorithm

A pseudo-code for iterative noise cancelation procedure is based on the mathematical background described in Sect. 3. In the pseudo-code, the function $\mathtt{LUT}(x, y)$ composes the look-up table by comparing symbols x and y.

Algorithm 1 The iterative noise cancelation algorithm

Initialization:
set the step-size μ_v
$\mathbb{T}_n\left(Q\left(|y(n)|\right), 1\right) \leftarrow \mathbb{F}^{-1}(n)$

Execution:
for all $k = 1, \cdots, l-1$ **do**
  $\mathbb{T}_n\left(Q\left(|y(n)|\right), k\right) \leftarrow \mathtt{LUT}(y(n), \hat{z}(n,k))$
  $\hat{z}(n,k) \leftarrow \mathbb{T}_n\left(Q\left(|y(n)|\right), k\right)$ ▹ eqn. (6)
  $\tilde{v}(n,k) \leftarrow z(n) - \hat{z}(n,k)$ ▹ eqn. (11)
  $\hat{z}(n,k+1) \leftarrow z(n) - \mu_v\tilde{v}(n,k)$ ▹ eqn. (13)
end for
return $\hat{z}(n) \leftarrow \hat{z}(n,l)$

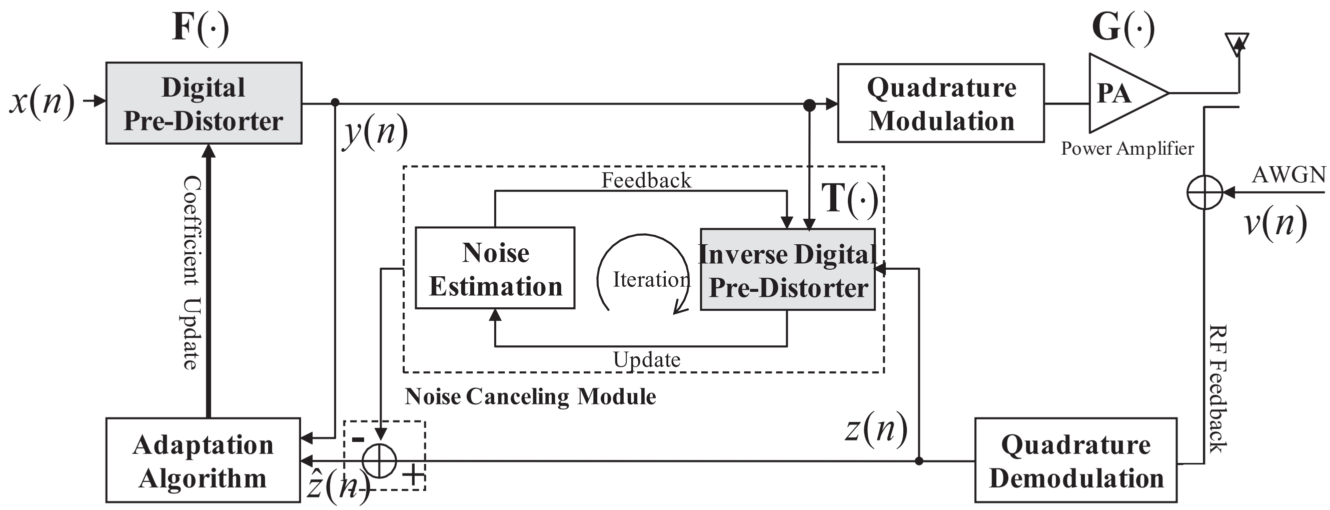

Fig. 2 Block diagram of the adaptive DPD with an iterative noise canceling module. (the dotted rectangle)

3. Convergence of the Algorithm Combined with Iterative Noise Cancelation

3.1 Signal Model

The modulating complex baseband input signal $x(n)$ and the predistorter output $y(n)$ can be written as

$$x(n) = r(n)e^{j\phi(n)} \tag{3}$$

$$\begin{aligned} y(n) &= \mathbb{F}(x(n)) \\ &= q[r(n)]\, e^{j(\phi(n)+\theta[r(n)])} \end{aligned} \tag{4}$$

where the functions $q[\cdot]$ and $\theta[\cdot]$ are to be determined by the characteristics of DPD, a gain-based LUT $\mathbb{F}(\cdot)$. Then, the HPA output signal is represented as

$$\begin{aligned} z(n) &= \mathbb{G}(y(n)) + v(n) \\ &= u[q[r(n)]]\, e^{j\left(\phi(n)+\theta[r(n)]+\Phi\left[q[r(n)]\right]\right)} + v(n) \\ &= \left(u[q[r(n)]] + \varepsilon_g\right) e^{j\left(\phi(n)+\theta[r(n)]+\Phi\left[q[r(n)]\right]+\varepsilon_p\right)} \end{aligned} \tag{5}$$

where $v(n)$ is the additive white Gaussian noise so that ε_g and ε_p for inevitable magnitute and phase error occurs, respectively, and the functions $u[\cdot]$ and $\Phi[\cdot]$ are to be determined by the characteristics of HPA $\mathbb{G}(\cdot)$.

The HPA model output $\hat{z}(n,k)$ in the noise canceling module can be obtained as

$$\begin{aligned} \hat{z}(n,k) &= \mathbb{T}_n\left(Q\left(|y(n)|\right), k\right) \\ &= \tilde{u}[q[r(n)]]\, e^{j\left(\phi(n)+\theta[r(n)]+\tilde{\Phi}\left[q[r(n)]\right]\right)} \end{aligned} \tag{6}$$

where the functions $\tilde{u}[\cdot]$ and $\tilde{\Phi}[\cdot]$ are to be determined by the characteristics of modeled HPA $\mathbb{T}_n(\cdot)$, k represents the kth iteration stage in the noise canceling module, $Q(|y(n)|)$ is the quantized amplitude of the predistorted signal $y(n)$, and $\mathbb{T}_n(Q(|y(n)|), k)$ is the temporal LUT used in the noise canceling module by comparing $\hat{z}(n,k)$ in (13) with $y(n)$. For initialization, $\mathbb{T}_n(Q(|y(n)|), 1)$ is set to the inverse function of DPD $\mathbb{F}^{-1}(n)$.

3.2 Steady-State Condition

In order to analyze the adaptive algorithm, it is assumed that the DPD coefficients are computed based on minimum squared error (MSE) criterion.

The error estimation between the actual HPA output and the estimated output is defined as

$$e(n) = |\hat{z}(n) - x(n)|^2 \tag{7}$$

Since the MSE function is a quadratic function, there exist iterative techniques to find the minimum. The iteration of the algorithm such as LMS (Least Mean Square) and RLS (Recursive Least Square) is terminated when the direction of the gradient becomes zero, i.e., $\nabla e(k) = 0$ for some k.

3.2.1 Magnitude Update (AM/AM)

The magnitude error $e_{\text{gain}}(n)$ can be defined as

$$e_{\text{gain}}(n) = |u[q[r(n)]] + \varepsilon_g - r(n)|^2. \tag{8}$$

where ε_g is an error due to additive noise $v(n)$.

By taking the derivatives with respect to q,

$$\begin{aligned} &\nabla_q e_{\text{gain}}(n) \\ &\quad= \frac{\partial}{\partial q}|u[q[r(n)]] + \varepsilon_g - r(n)|^2 \\ &\quad= 2|u[q[r(n)]] + \varepsilon_g - r(n)| \cdot u'[q[r(n)]] \cdot q'[r(n)] \end{aligned} \tag{9}$$

where $\frac{\partial}{\partial q} q[\cdot] = q'[\cdot]$, and the chain rule of the derivative is used because of the composites of two functions.

Lemma 1 (Steady-State Condition): The magnitude update of DPD will stop when DPD is equal to the inverse function of HPA, $q[r(n)] = u^{-1}[r(n)]$.

Proof: The stopping criterion is obtained from (9) based on $\nabla_q e_{\text{gain}}(n) = 0$ as

$$u\left[q\left[r(n)\right]\right] = r(n).$$

By taking inverse function of u

$$q\left[r(n)\right] = u^{-1}\left[r(n)\right].$$

Because ε_g^2 cannot be estimated and removable, the residual MSE is equal to ε_g^2. ■

3.2.2 Phase Update (AM/PM)

Similarly, the phase error $e_{\text{phase}}(n)$ can be defined as

$$e_{\text{phase}}(n) = \left|\, \theta\left[r(n)\right] + \Phi\left[q\left[r(n)\right]\right] + \varepsilon_p \,\right|^2 \tag{10}$$

where ε_p is error due to additive noise $v(n)$.

By taking the derivatives with respect to θ,

$$\begin{aligned}\nabla_\theta e_{\text{phase}}(n) &= \frac{\partial}{\partial\theta}\left|\, \theta\left[r(n)\right] + \Phi\left[q\left[r(n)\right]\right] + \varepsilon_p \,\right|^2 \\ &= 2\left|\, \theta\left[r(n)\right] + \Phi\left[q\left[r(n)\right]\right] + \varepsilon_p \,\right| \cdot \theta'\left[r(n)\right].\end{aligned}$$

Lemma 2 (Steady-State Condition): The phase update of DPD will stop when DPD is equal to the phase distortion caused by HPA, $\theta\left[r(n)\right] = -\Phi\left[q\left[r(n)\right]\right]$.

Proof: The stopping criterion is obtained from (11) based on $\nabla_\theta e_{\text{phase}}(n)$ as

$$\theta\left[r(n)\right] + \Phi\left[q\left[r(n)\right]\right] = 0.$$

Note that the residual MSE is equal to ε_p^2. ■

3.3 Convergence of Iterative Noise Cancelation

The estimated noise $\tilde{v}(n,k)$ in the noise canceling module can be calculated as

$$\tilde{v}(n,k) = z(n) - \hat{z}(n,k). \tag{11}$$

Since $\mathbb{T}$ is based on inverse LUT $\mathbb{F}^{-1}$, it follows that $\tilde{u}[\cdot]$ and $\tilde{\Phi}[\cdot]$ are asymptotically converged from *Lemma 1* and *Lemma 2* as

$$\begin{aligned}&\tilde{u}\left[r(n)\right] = q^{-1}\left[r(n)\right] \to u\left[r(n)\right], \\ &\tilde{\Phi}\left[q\left[r(n)\right]\right] = -\theta\left[r(n)\right] \to \Phi\left[q\left[r(n)\right]\right].\end{aligned} \tag{12}$$

where $\to$ means "convergence almost surely." Since the accuracy of estimating noise is increased by iteratively canceling the noise, $\tilde{v}(n,k)$ approaches the actual noise $v(n)$.

The output of iterative noise cancelation $\hat{z}(n)$ can be obtained by

$$\hat{z}(n) = z(n) - \mu_v \tilde{v}(n,k) \to \mathbb{G}\left(y(n)\right) \tag{13}$$

where μ_v is the step-size for adaptive update which adjusts the convergence speed. Note that by subtracting the noise, it converges into the output of $\mathbb{G}\left(y(n)\right)$ which is the output of actual HPA without effect of noise, $\varepsilon_g \to 0$ and $\varepsilon_p \to 0$. Using $\hat{z}(n)$ in (6), more accurate HPA estimation is possible, which leads to larger out-of-band suppression.

Using (13), the temporal LUT for next iteration in the noise canceling module can be renewed into $\mathbb{T}_n\left(Q\left(|y(n)|\right), k+1\right)$. Repeating the procedures from (6) to (13) until the pre-specified maximum iteration value l, the output signal with iterative noise cancelation $\hat{z}(n) = \hat{z}(n,l)$ consequently obtained and input to the adaptation algorithm block in order to compose DPD $\mathbb{F}(\cdot)$.

4. Simulation Results

Simulation results are given to demonstrate how the proposed DPD technique could improve the performance in conjunction with the iterative noise cancelation. AWGN is regarded as the noise in the system.

As an HPA model in Fig. 3, we used a popular TWTA model specified in Saleh's original work [9]. As a transmit system, an ATSC system with 8-VSB modulation was applied, and the 256-entry LUT with uniform spacing was employed. MSE was used as a criterion to evaluate the performance of the DPD method such as convergence rate and constellation quality.

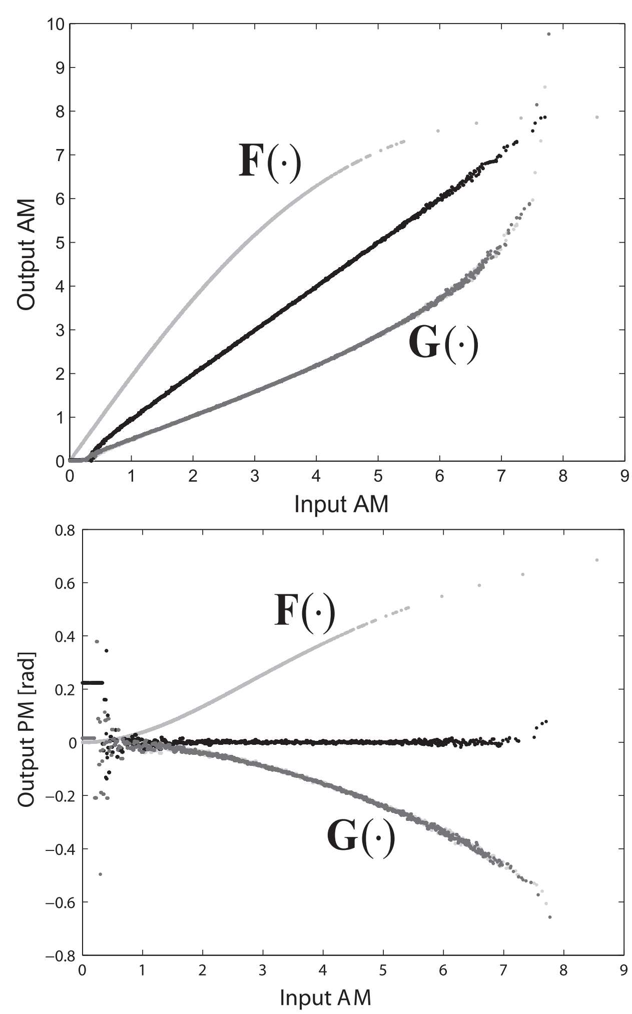

Fig. 3 (top) AM-AM (bottom) AM-PM graph in DPD with and without INC (SNR=15 dB).

4.1 Adaptation Capability: MSE

Figure 4 shows the MSE performance during the adaptation procedure as a function of SNR. For obtaining MSE value at each iteration, $E_n[e(n)]$, 50 ATSC segments are averaged. The step-size μ_v is set to 0.1 with 25 iterations. As the SNR increases, the MSE is decreased at the steady state. Compared to the conventional method without iterative noise cancelation, an additional decrease in the error floor can be achieved by applying the iterative noise cancelation. The difference amount of MSE between with and without INC is smaller as SNR increases. That is, the effectiveness of INC gains at lower SNR, i.e. 5 dB, as shown in Fig. 4.

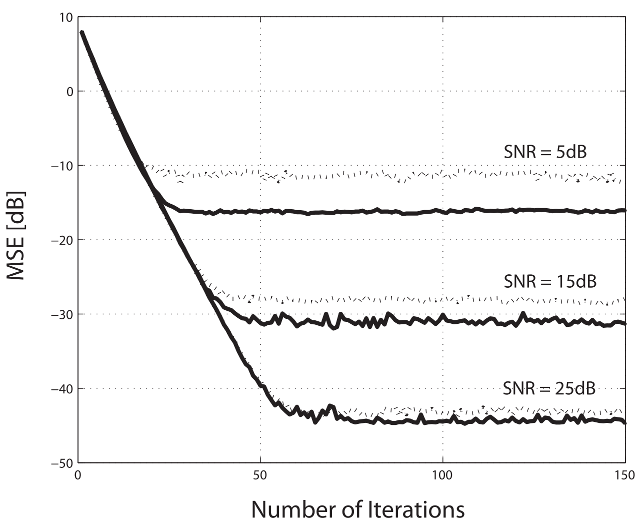

Fig. 4 Mean-square error performance during the adaptation procedure according to SNR of RF feedback loop ($\mu_v = 0.1$) with (solid) and without (dotted) INC.

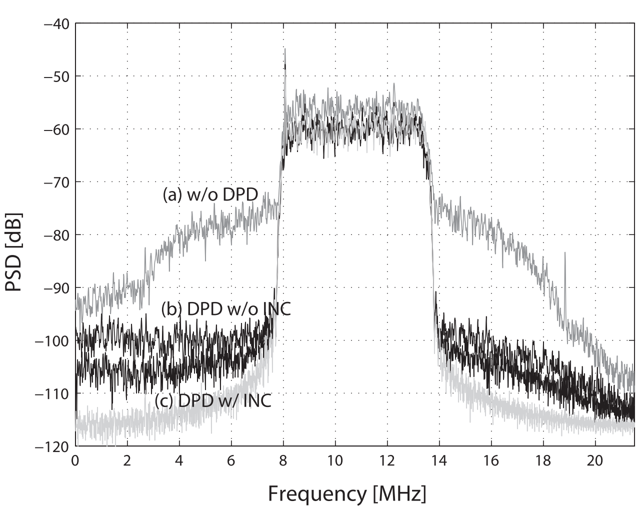

Fig. 5 Effectiveness of PD in suppressing spectral regrowth (a) Output spectrum without PD (b) Output spectrum with PD only (c) Output spectrum with PD and iterative noise cancelation. (SNR=15 dB).

4.2 Intermodulation Distortion: PSD

In the simulation, the step size for the adaptation is set to $\mu_v = 0.1$. Figure 5 illustrates the output spectrum with and without predistorter in comparison to the input signal spectrum. The non-linear amplifier causes spectral regrowth as well as in-band distortion. With predistorter, it is clearly demonstrated that approximately 20∼25 dB improvement in out-of-band suppression is obtained. A predistorter with an iterative noise cancelation additionally suppresses the out-of-band emission by approximately 5 dB.

5. Experimental Validation

The designed predistorters, simulated in Sect. 4, were used to linearize the power amplifier under test. Figure 6 depicts a block diagram of the experimental setup. In order to validate the proposed DPD-INC technique, reference ATSC DTV exciter and Tektronix RFA300a [12] were used for the test. PA linearization performances in conjunction with iterative noise cancelation by means of the gain-based LUT were investigated on the 8-VSB ATSC DTV system. The signal has a 6-MHz bandwidth and the mask filter was not applied.

The developed device has two 512-entry AM/AM and AM/PM LUTs programmed in MATLAB using both with and without iterative noise cancelation algorithms. Actual 100 Watt PA is used, and the linearization capabilities were evaluated only for the memoryless nonlinear characteristics of the amplifier.

Note that RFA300a is used as a reference instrument for measurement standards and approving the radio station.

Figure 7 illustrates the spectrum of HPA output signal captured by RFA300a analyzer. The solid (red) line represents the FCC DTV emission mask [13]. Since the spectrum can be easily suppressed by using DTV mask filter, suppressing within the first 500 kHz from the authorized channel edge is important in order to meet the regulation mask. From this perspective, the use of a predistorter with iterative noise cancelation suppresses the out-of-band emission by approximately 2∼3 dB with a breakpoint at 500 kHz.

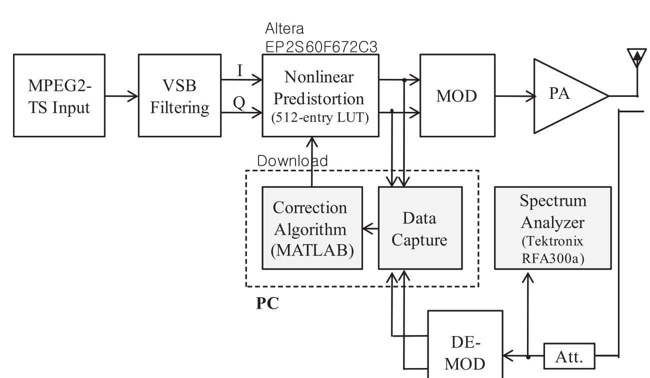

Fig. 6 Block diagram of experimental setup for PA linearization test.

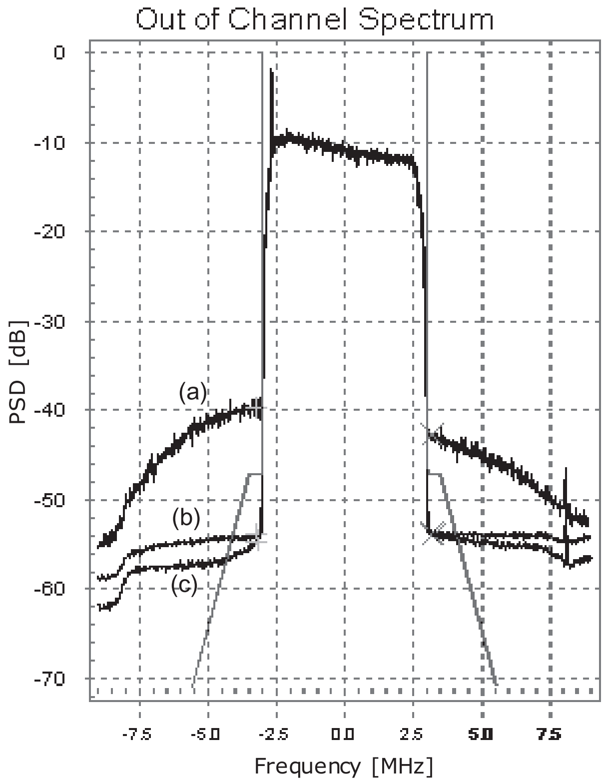

Fig. 7 Measured 8-VSB ATSC DTV signal spectra with and without the iterative noise cancelation (a) without linearization (b) DPD without INC (c) DPD with INC.

In practice, as the DPD circuit becomes older owing to long operation, the SNR of the RF feedback loop steadily decreases as the DPD becomes older. Hence, as shown in Fig. 4, the benefit of INC increases as the DPD circuit is aged.

6. Conclusion

In this paper, an improved adaptive estimation technique of HPA nonlinear characteristics has been proposed based on the minimization of the mean square error of HPA AM/AM and AM/PM conversion priorities by introducing iterative noise cancelation. Compared to the conventional method without iterative noise cancelation, an additional decrease in the error floor can be achieved by applying the iterative noise cancelation. With predistorter, it is clearly demonstrated that approximately 2∼3 dB additional suppression in out-of-band suppression is obtained, and this gain was also verified by a hardware implementation.

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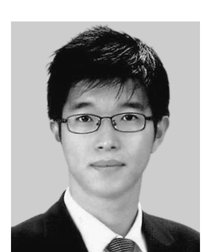

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