Effects of Anti-Aliasing Filters in Feedback Path of Adaptive Predistortion

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***Abstract* -- Adaptive predistortion is one of the most promising methods to linearize RF power amplifiers (PAs). Its linearity performance is, however, limited by various memory effects existing in the system. In this paper, we study the effects of anti-aliasing filters in the feedback path of adaptive predistortion. Different filter approximations are examined with a class AB power amplifier by means of simulations. Of the filters studied, Butterworth filters perform better than Chebyshev filters when they have the same noise-equivalent bandwidth (NEB). Simulation results also show that the cutoff frequency of the anti-aliasing filter affects both the convergence speed of the adaptation algorithm and the level of mean square error (MSE) between the desired and the feedback signals. To a certain degree the higher the cut-off frequency of the filter is, the less the number of iterations is needed by the adaptation process.**

I. Introduction

During the past decade, substantial attention has been devoted to developing techniques to make highly efficient and highly linear power amplifiers (PA). But from a traditional PA design point of view, the efficiency and linearity are opposite requirements. In order to achieve both, we have to apply linearization techniques to a power efficient but nonlinear amplifier. Among the proposed linearization techniques, predistortion is a powerful method due to its wideband operation capability and simplicity [1][2]. An input-output distortion free system can be obtained by predistorting the input signal according to the inverse of the amplifier's nonlinearities. By applying an adaptive process to the predistortion method, stable compensation of nonlinear distortion can be obtained in spite of large variations in operating conditions of the amplifier, e.g. temperature, load, and supply voltage etc. In theory, intermodulation distortion (IMD) generated by a nonlinear PA can be fully cancelled. However, a practical system always suffers from more or less errors and memory effects, which cause non-ideal cancellation. In the published papers, many of these effects are identified and analyzed, such as the misalignments of the quadrature modulators [3][4], feedback delay error [5] and the reconstruction filters[6][7].

Filters contribute significant memory effects in predistortion systems [6]-[8]. Although the imperfections of the anti-aliasing filters in the feedback path for adaptive predistortion do degrade the performance of the system [8], their effects so far have not been treated in the literature to the authors' knowledge. In this paper we investigate this problem by means of analysis and simulations. A guideline for designing the feedback filters for an adaptive predistortion system is given.

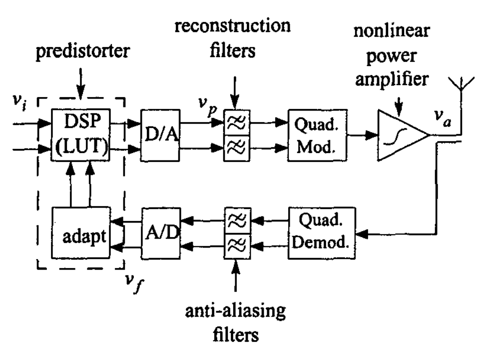

Fig. 1. Adaptive predistortion linearizer.

II. System Description

An adaptive predistortion system is shown in Fig. 1. All signals are treated as complex baseband, and denoted by lower case letter v in time domain and upper case letter V in frequency domain, with a subscript to identify the location in the system as illustrated in Fig. 1.

The forward path consists of a complex gain based predistorter, D/A converter, reconstruction filters, a quadrature modulator upconverting the filtered baseband signals to RF and the power amplifier, modeled as a memoryless nonlinearity. The predistorter has the precise inverse transfer characteristic of the nonlinear amplifier

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and is implemented by a look-up-table (LUT) in Cartesian form, addressed by the input signal amplitude. In such a way, the whole forward path ideally achieves linear amplification with a gain K:

$$v_a = v_i F(|v_i|) G(|v_i F(|v_i|)|) = K v_i \quad (1)$$

where $F(|v|)$ and $G(|v|)$ are the complex gain of the predistorter and the amplifier respectively.

Adaptation of the predistorter is achieved by comparing the amplifier output v_f with the desired linear output Kv_i and updating the values in LUT to minimize their difference

$$v_{fe} = v_f - K v_i \quad (2)$$

The adaptation algorithm used in this work is a simple linear scheme with batch processing [7]. That means when there are M samples of the v_i and v_f stored in the memory, samples with the same amplitude are collected and the average error vector is computed. Thus the LUT is updated as follows:

$$F(|v_i|, k+1) = F(|v_i|, k)\left(1 - \frac{\delta}{N} \sum_{n=1}^{N} \frac{v_{f,n} - K v_{i,n}}{v_{f,n}}\right) \quad (3)$$

where $F(|v_i|, k)$ is the LUT entry at iteration k and address $|v_i|$, δ is the convergence factor, N is the number of samples with the same amplitude in the M point buffer and the samples are indexed by n. Anti-aliasing filters will, however, interfere with this adaptation process.

The normalized passband transfer function of a nonideal filter may be modeled as:

$$H(f) = 1 + D(f) \quad (4)$$

where D is a complex quantity and a function of frequency, representing both the amplitude and phase ripple of the filter. We assume the passband of the filter is wide enough to pass the signal v_i and there are no other error sources in the system. Then the feedback signal v_f with the effects of the filters can be derived as:

$$v_f = v_a + v_a \otimes d \quad (5)$$

where '$\otimes$' denotes the convolution operation and d is the time domain function of D. Substituting (5) into (3), a new updating equation for the LUT can be obtained:

$$F(|v_i|, k+1) = F(|v_i|, k)\left(1 - \frac{\delta}{N} \sum_{n=1}^{N} \frac{v_a + v_a \otimes d - K v_{i,n}}{v_a + v_a \otimes d}\right) \quad (6)$$

It's clear from (6) that if $d \neq 0$, it will affect the convergence of the adaptation and cause the feedback signal to deviate from the desired one. Since this deviation is a function of frequency, the adaptive process cannot compensate for this error. Therefore, this memory effect will introduce out-of-band spectral products leading to adjacent channel interference (ACI).

III. Simulation Results

In this section the simulations are carried out by exercising the system with a $\pi/4$ shifted QPSK modulation, square-root raised cosine filtered with roll-off factor 0.35. A class AB power amplifier is used and its characteristics are shown as in Fig. 2. The LUT table size is set to 64. The convergence factor of the adaptation algorithm is set to 0.1 which gives a good compromise between averaging of noisy perturbation and rapid adaptation. The purpose of the anti-aliasing filters is to keep the noise level down. For practical reasons, lower order filters are preferred and also higher order filters cannot provide much improved noise suppression. In this study, only third-order Butterworth and Chebyshev filters are considered.

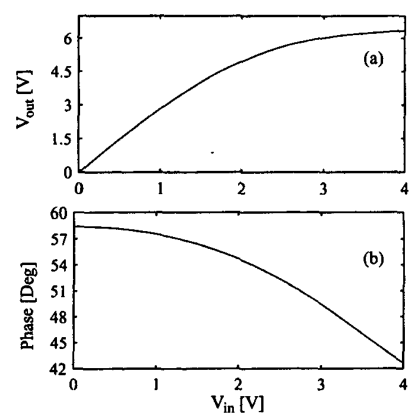

Figure 2. AM-AM (a) and AM-PM (b) characteristics of a class AB power amplifier

First, the effects of the filters' cut-off frequency are studied. The cut-off frequency is defined as the frequency at which the filter attenuation has reached 3dB for Butterworth filters, and it is the frequency at which the filter attenuation has reached the level corresponding to the maximum ripple of the filter for the Chebyshev filters. These frequencies are normalized to the symbol rate of the modulated signal. Third-order Butterworth filters with four different cut-off frequencies are used as anti-aliasing filters in the system. Figure 3 gives the simulated output spectra, where for each filter the parameters in the parentheses indicate its order and cut-off frequency, respectively. Although lower cut-off frequency is preferred from a noise level point of view, it will result in higher ACI, see Fig. 3. However, when the cut-off frequency is high enough a further increase will not noticeably reduce the ACI, which is now mainly limited by other facts such as the size of the LUT. Thus the cut-off frequency of the filter should be kept as low as possible while not affecting the ACI level.

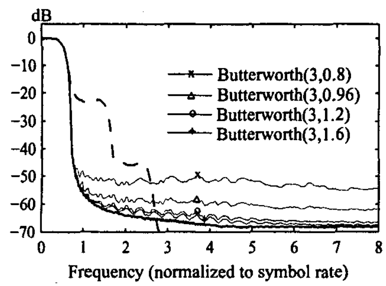

Figure 3. The output spectra of the system for four Butterworth anti-aliasing filters with different cut-off frequencies.

The convergence speed of the adaptation algorithm as a function of the iterative number is investigated by calculating the total mean squared error (MSE) between the desired and the actual feedback signals [9]. Denoting the modulation amplitude probability density function as $p(v_i)$, the total MSE becomes:

$$\sigma_a^2 = \int_0^{\infty} \left|v_i F(|v_i|) G(|v_i F(|v_i|)|) - v_i\right|^2 p(v_i) dv_i \qquad (7)$$

Figure 4 shows the MSE between the desired and the feedback signals as a function of adaptation iterations for the four filters. Since the MSE also includes in-band error which does not affect the adjacent channels, the level of MSE is higher than the level of ACI shown in Fig. 3. It is readily seen that both the achievable MSE level and the convergence speed of the adaptation algorithm is affected by the cut-off frequency of the anti-aliasing filters. To an achievable MSE level, less iteration numbers are required for a filter with higher cut-off frequency. Also, lower filter cut-off frequency will result in higher residual MSE which can not be reduced further by increasing the number of iterations, see Fig. 4.

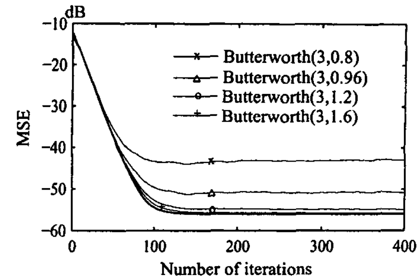

Figure 4. The corresponding MSE versus numbers of iterations for various Butterworth filters.

Next, the effects of ripple inside the filter passband are investigated. Four different filters with the same noise-equivalent bandwidth (NEB) are treated as anti-aliasing filters. The definition of NEB [10] is given as:

$$B_{neq} = \frac{\int_{-\infty}^{+\infty} |H(f)|^2 df}{2H_{max}^2} \qquad (8)$$

where H_{max} denotes the maximum value of $|H(f)|$ in the passband of the filter. The four filters used are one third-order Butterworth filter and three third-order Chebyshev filters with in band ripple level 0.1dB, 0.2dB and 0.5dB, respectively. It is known from Fig. 3 and 4 that the effects of the Butterworth filter are negligible when its cut-off frequency is 1.6 times the signal symbol rate. Thus, it is set as a reference for determining the cut-off frequency of the other three filters such that they all have the same NEB. The resulted three Chebyshev filters are given in Fig. 5, where the parameters in each parenthesis indicate the filter's order, ripple level, and cut-off frequency, respectively. It is worth noticing that they all have attenuation less than 3dB within the frequency band below 1.6 times the signal symbol rate. Although the filters have both phase and amplitude ripple inside their passband, the effects caused by amplitude ripple are dominant. This claim is certified by simulations. The major reason is that both Butterworth and Chebyshev filters have a nearly linear phase response over about three-fourths of the passband, in which most of the signal energy is located. Butterworth filter has maximum flat amplitude response in the signal band and 0.5dB ripple Chebyshev filter has the sharpest transition slope. Figure 5 gives the simulated output spectra with respect to the four filters. It is clearly seen that the Butterworth filter has the best performance among these four filters and the lower-ripple Chebyshev filter has better performance than the higher-ripple one. This result indicates that the amplitude ripple of the filter inside the signal band can significantly affect the performance of the predistortion system. The same conclusion can also be drawn based on the MSE level shown in Fig. 6. Due to the larger ripple inside the signal band for the Chebyshev filters, the MSE is much higher compared with its corresponding ACI level. As with the effects of the filter's cut-off frequency, the MSE level cannot be reduced further even with more iterations, as seen in Fig.6.

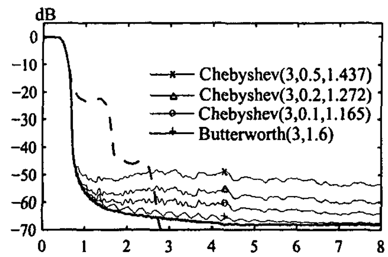

Figure 5. The output spectra of the system for four anti-aliasing filters with the same NEB but different in-band ripple.

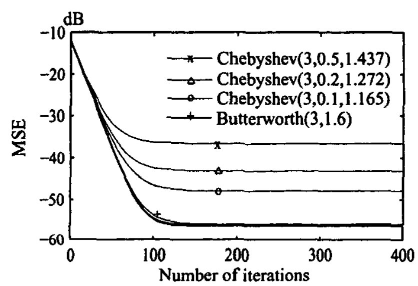

Figure 6. The corresponding MSE versus numbers of iterations for the filters used in Fig. 6.

Note that the results without predistortion as well as with predistortion having ideal anti-aliasing filters are also plotted in Fig. 3, 4, 5 and 6 using dashed and solid emphasis line, respectively.

IV. CONCLUSIONS

The effects of imperfections of the feedback filters on an adaptive predistortion system for linearizing RF power amplifier have been studied. The obtained results show that the memory introduced by the filters affects the convergence speed of the adaptation algorithm and the residual ACI level. The lower cut-off frequency the filter has, the more iterations the adaptation requires to reach an achievable MSE level. However, when the cut-off frequency is high enough a further increase will not noticeably reduce the ACI, which is now limited by other facts such as table size. For a given NEB level, the filters with larger amplitude ripple within the signal band will introduce more out-of-band spectral products which lead to higher ACI levels. The degradation of system performance due to the memory effects of feedback filters can not be counteracted with more adaptive iterations. Therefore, such imperfections must be taken into account when designing an adaptive predistortion linearizer. Otherwise, they may cause a well designed system fail to meet the linearity requirements.

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