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Performance comparison of adaptive digital predistortion algorithms using adaptation time for mobile WiMAX power amplifier applications

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Abstract: In this paper, we propose that adaptation time be considered a parameter for evaluating the performance of digital filter algorithms in adaptive digital predistorted high-power amplifiers. An adaptive predistorted high-power amplifier with an average power of 37.5 dBm was designed and fabricated for mobile worldwide interoperability for microwave access (WiMAX) test signals. For performance comparisons using adaptation time, several effective algorithms (such as the least-mean-squares (LMS), recursive-least-squares (RLS), and affine projection (AP) algorithms) were implemented. According to the experimental results, the RLS and AP algorithms achieved a similar linearity improvement of 11 dBr. However, the AP algorithm had the advantage of a faster adaptation time than the other algorithms.

Key words: High power amplifier, linearization, digital predistortion, adaptive filter algorithm, mobile worldwide interoperability for microwave access

1. Introduction

In modern communication systems that use digital modulation signals, adaptive digital predistortion is used as one of the attractive linearity enhancement technologies for high-power amplifiers. Digital predistortion (DPD) can maintain the optimized linearity by adapting the coefficients of DPD to the variations of the high-power amplifier characteristics due to internal and external environmental changes. In particular, adaptive DPD with an indirect learning structure that considers memory effects can guarantee the high linearity performance of high-power amplifiers [1,2].

The performance of adaptive DPD is affected by the adaptive filter algorithms in the predistortion trainer. Several adaptive digital filter algorithms, such as the least-mean-squares (LMS), recursive-least-squares (RLS), and affine projection (AP) algorithms, have been used in DPD [3–5]. There are also various adaptive filter algorithms, but these algorithms are mostly modified forms of the LMS, RLS, and AP algorithms [5–7]. The performance characteristics of adaptive digital predistorters with these filter algorithms can be evaluated by measuring the linearity improvement and adaptation speed. However, most studies have been carried out to evaluate the improvement in linearity. There has also been some research into reducing the computational complexity and improving the convergence rate of the DPD. However, these parameters cannot provide direct information about the adaptation speed of the adaptive algorithms, even though it is one of the most important performance evaluation parameters for DPD [2–8].

In this paper, adaptation time was used as a performance parameter for evaluating adaptive DPD. For

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the performance comparisons of the adaptation time, the LMS, RLS, and AP algorithms were selected because these algorithms are standard algorithms for the modified adaptive filter algorithms.

2. Adaptation speed

Figure 1 shows the block diagram of an adaptive digital predistorted power amplifier (DPDPA). A digital predistorter linearizes the nonlinear response of a high-power amplifier over an operating region by predistorting a baseband signal prior to modulation, up-conversion, and amplification. A digital predistorter consists of a finite impulse response (FIR) filter with polynomial order of p and memory length of m. The coefficients $W_{p,m}$ of the predistorter are updated by using adaptive filter algorithms such as the LMS, RLS, and AP algorithms. The operating characteristics of high-power amplifiers are influenced by external environmental changes or rising temperature, due to the effect of heat on the internal active devices. These variations of the operating characteristics cause the coefficients of DPD to be incorrect and deteriorate the linearity performance of the DPDPA.

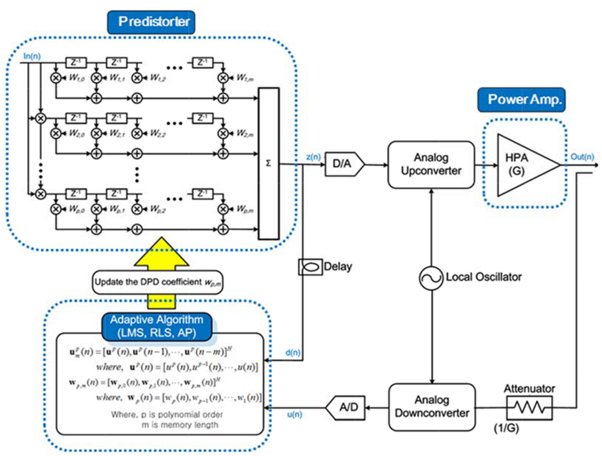


Figure 1. Block diagram of an adaptive digital predistorted power amplifier.

The DPDPA tries to maintain optimized linearity by continuously updating the coefficients of DPD. However, it takes time to recover the optimized linearity of adaptive DPDPA.

Adaptive filter algorithms in DPD are generally evaluated in terms of their error performance, convergence rate, and computational complexity [9–11]. While error performance is related to the linearity improvement of adaptive DPD, convergence rate and computational complexity affect adaptation speed. Adaptation speed represents how quickly the filter coefficients are updated to the optimized values and is affected by the convergence rate and computational complexity of the adaptive filter algorithm, respectively [12]. However, adaptation speed has not been used as a performance evaluation parameter, because it might not be quantitatively related. Heretofore, the convergence rate or computational complexity of the adaptive filter algorithm has been used as a characteristic parameter of the adaptive filter algorithm. For quantitative performance evaluation of the adaptation speed, it is necessary to define an evaluation parameter that takes both the convergence rate and computational complexity into account.

In this paper, adaptation time is proposed to quantitatively represent adaptation speed in DPD as a function of the number of updates to the digital predistortion coefficients and iteration numbers of the coefficients, and the computational complexity of the adaptive filter algorithm:

$$t = n_1 \times n_2 \times \chi \times \tau_m \quad (1)$$

where n_1 is the number of updates to the digital predistortion coefficients, n_2 is the iteration number of the coefficients in the adaptive filter algorithm, χ is the computational complexity of the adaptive filter algorithm (which is defined as the number of multiplications required for one operation in the adaptive filter algorithm), and τ_m [s] is the time required to calculate one multiplication.

As shown in Eq. (1), adaptation time has a relatively different value depending on not only computational complexity but also iteration number of the coefficients in the adaptive filter algorithm. Furthermore, the effect of the number of updates to the digital predistortion coefficients on the adaptation time is considered.

3. Performance analysis

In order to evaluate the adaptation speed of various adaptive filter algorithms (LMS, RLS, and AP) using the proposed adaptation time, a high-power amplifier was designed and fabricated at a frequency of 1867.5 MHz with 5 MHz bandwidth for the mobile WiMAX test signal. The power amplifier had an average output power of 37.5 dBm, gain of 42.1 dB, and ACPR of 41.7 dBr at an offset frequency of ± 4.77 MHz.

For the performance comparison of the filter algorithms in DPD, a mixed software–hardware design method called a "connected solution" was used by combining the electronic design automation software with the testing and measurement equipment.

Figure 2 shows the test setup for the connected solution. There were three main measurement steps using the connected solution. In the first step, the WiMAX test signal was generated in the advanced design system from the personal computer (PC) and was downloaded into an arbitrary waveform generator. The power amplifier (PA) with a peak envelope power of 47 dBm amplified the radio frequency (RF) signal from the vector signal generator (VSG) and the amplified signal went into the spectrum analyzer after attenuation. In the second step, the output samples collected from the spectrum analyzer and the samples downloaded from the PC were first delay-matched and compared to estimate the coefficient sets for the FIR filter in MATLAB. The predistorted signal from the PC was downloaded into the VSG in the third step. This loop was repeated until the algorithm converged.

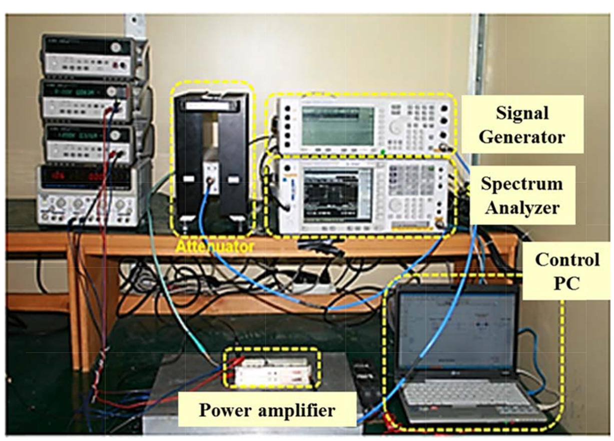


Figure 2. The test setup for the connected solution.

Error performance was calculated by normalizing the mean square error (NMSE), as shown in Eq. (2), where the desired signal d(n) is the PA input signal and y(n) is the output signal of the predistortion trainer.

$$NMSE(dB) = 10\log_{10}\left(\frac{\sum_{n=1}^{N}|d(n)-\hat{y}(n)|^2}{\sum_{n=1}^{N}|d(n)|^2}\right) \tag{2}$$

For the test, the seventh-order polynomial series with a memory length of two was extracted for the algorithms.

Figure 3 shows the calculated convergence rate of each algorithm. The LMS algorithm converges to an NMSE of approximately –38.5 dB after 60,000 iterations, while the RLS converges to an NMSE of –40 dB after 5000 iterations. In the case of the AP algorithm, an NMSE of –38.5 dB was achieved after 10,000 iterations.

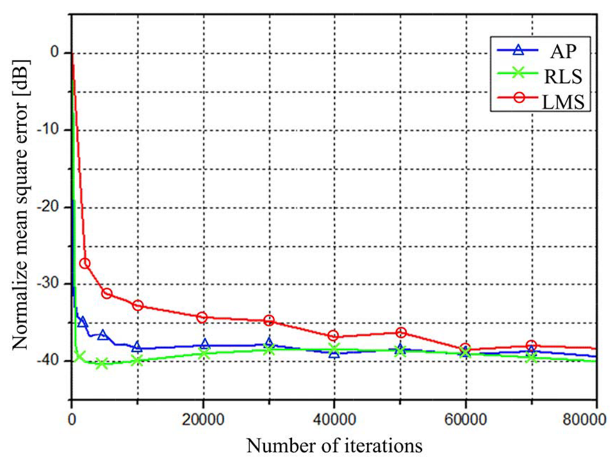
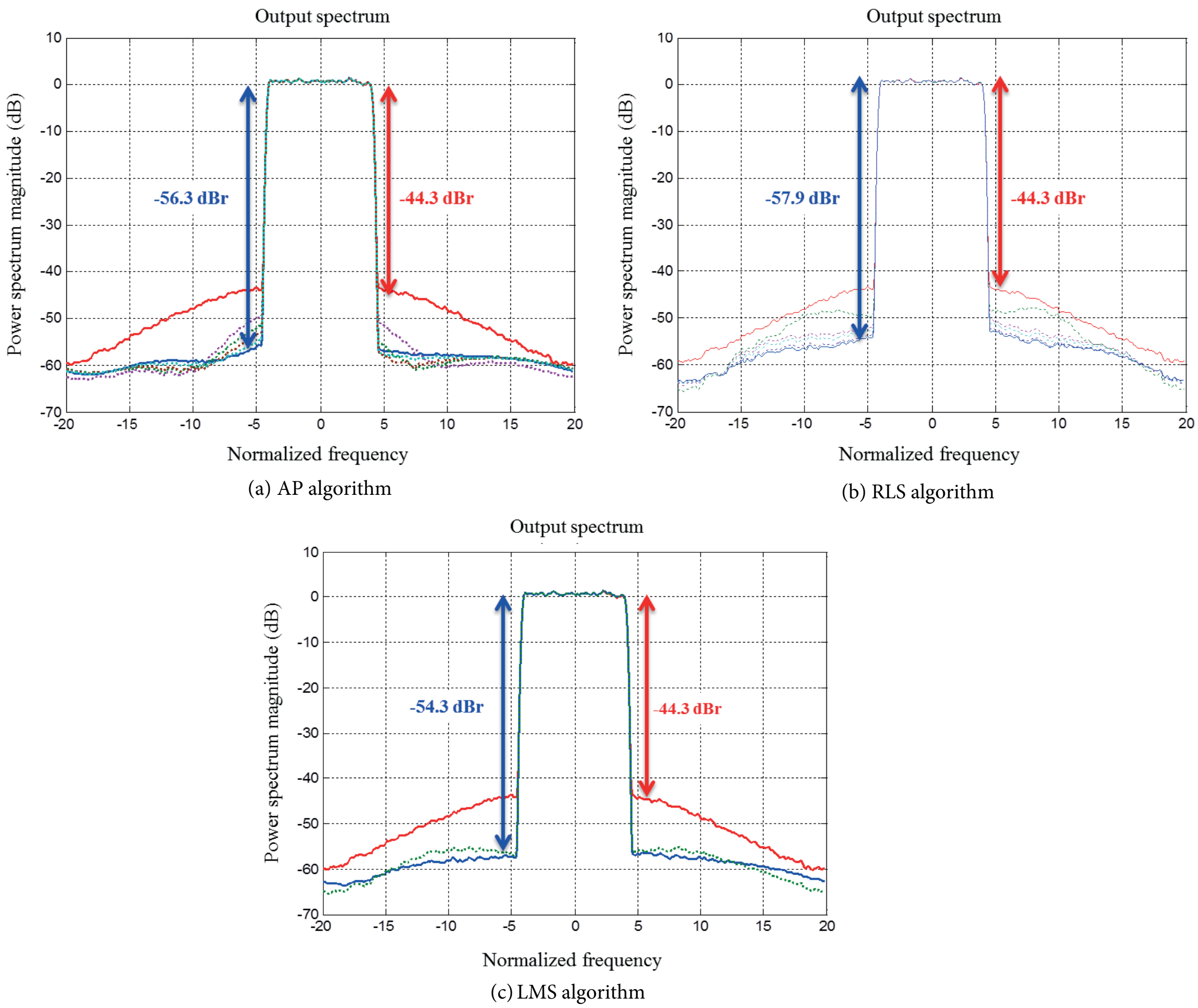


Figure 3. The calculated convergence rate of the RLS, AP, and LMS algorithms.

Figure 4 shows the simulated output spectrum of the adaptive digital predistorted power amplifier using the RLS, LMS, and AP algorithms. For the AP algorithm, 10,000 iterations of the adaptive filter algorithm, a computational complexity of 112, and 5 updates of the predistorter were required to achieve a lower adjacent channel power ratio (ACPR) of –56.3 dBr and upper ACPR of –57.3 dBr. An adaptation time of $5.56 \times 10^6\tau_m$ was calculated. The RLS algorithm used 5000 iterations, a computational complexity of 1764, and 2 updates with an adaptation time of $17.64 \times 10^6\tau_m$ for a lower ACPR of –57.9 dBr and upper ACPR of –57.5 dBr, while the LMS algorithm achieved a lower ACPR of –54.3 dBr and upper ACPR of –52.9 dBr for 60,000 iterations, a computational complexity of 43, and 5 updates with an adaptation time of $12.9 \times 10^6\tau_m$.

Table 1 shows a comparison of the simulation results of the performance parameters for the DPDPA based on these algorithms. The RLS algorithm had the fewest iterations and highest computational complexity. On the other hand, the most iterations and lowest computational complexity were obtained by using the LMS algorithm. Even though iteration number and computational complexity of the AP algorithm are in between those of the LMS and RLS algorithm, the AP algorithm obtained the shortest adaptation time among these algorithms. From the results, it is clear that the proposed adaptation quantitatively represents the adaptation speed of the adaptation filter algorithm. The RLS and AP algorithms show a good improvement in linearity.



(a) AP algorithm

(b) RLS algorithm

(c) LMS algorithm

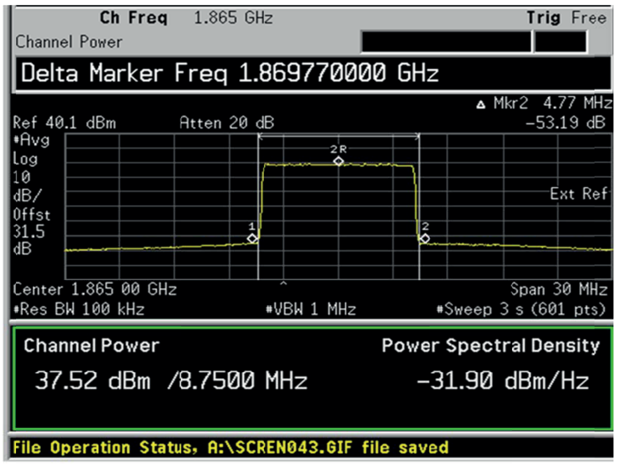
Figure 4. Simulated output spectra of adaptive digital predistorted power amplifiers based on the AP, RLS, and LMS algorithms.

Table 1. Simulated linearity improvements of adaptive filter algorithms.

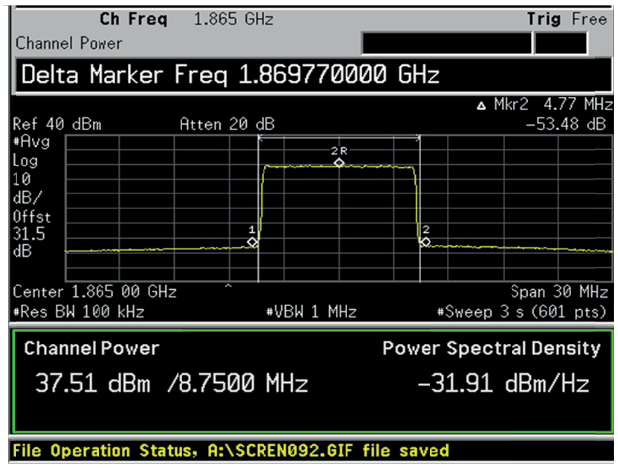
	PDCs update	Iteration number	Computational complexity	Adaptation time	Linearity upper/ lower (at –4.77 MHz)
AP	1	10,000	112	$1.12 \times 10^6 \tau_m$	–51.6/–50.2
	2			$2.24 \times 10^6 \tau_m$	–56/–52.2
	3			$3.36 \times 10^6 \tau_m$	–58.1/–53.7
	4			$4.48 \times 10^6 \tau_m$	–58.1/–56.2
	5			$5.56 \times 10^6 \tau_m$	–57.3/–56.3
RLS	1	5000	1764	$8.82 \times 10^6 \tau_m$	–56.6/–57.4
	2			$17.64 \times 10^6 \tau_m$	–57.5/–57.9
LMS	1	60,000	43	$2.58 \times 10^6 \tau_m$	–48.46/–49.78
	2			$5.16 \times 10^6 \tau_m$	–50.49/–52.03
	3			$7.74 \times 10^6 \tau_m$	–51.17/–52.69
	4			$10.32 \times 10^6 \tau_m$	–52.34/–53.81
	5			$12.9 \times 10^6 \tau_m$	–52.86/–54.31

PDCs: Predistorter coefficients.

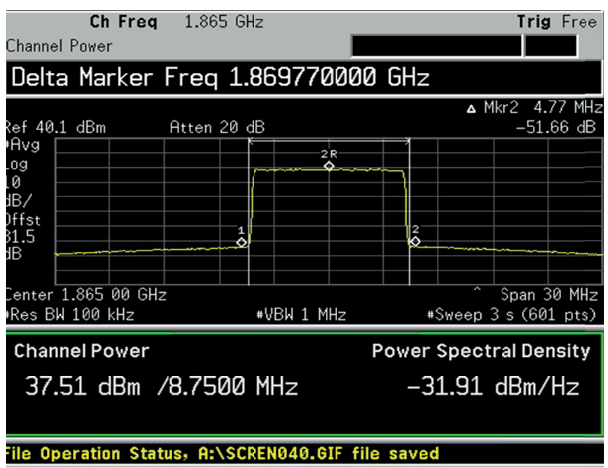
Figure 5 and Table 2 show the measured output spectrum and linearity improvement of the adaptive DPDPA using these algorithms, respectively.



(a) AP algorithm



(b) RLS algorithm



(c) LMS algorithm

Figure 5. Measured output spectra of adaptive digital predistorted power amplifiers based on the AP, RLS, and LMS algorithms.

Table 2. Measured linearity improvements of adaptive filter algorithms.

	PDCs update	Adaptation time	Linearity upper/lower (at –4.77 MHz)
AP	1	$1.12 \times 10^6 \tau_m$	–53.07/–52.84
	2	$2.24 \times 10^6 \tau_m$	–53.19/–53.13
RLS	1	$8.82 \times 10^6 \tau_m$	–53.48/–53.88
LMS	1	$2.58 \times 10^6 \tau_m$	–48.7/–49.76
	2	$5.16 \times 10^6 \tau_m$	–50.02/–50.38
	3	$7.74 \times 10^6 \tau_m$	–50.63/–51.13
	4	$10.32 \times 10^6 \tau_m$	–51.21/–51.53
	5	$12.9 \times 10^6 \tau_m$	–51.66/–52

PDCs: Predistorter coefficients.

It is clear that the experimental results are similar to the simulation results. However, the measured results indicate that an ACPR of –53 dBr is the maximum that can be achieved. From the measured results, the RLS algorithm shows the best linearity improvement, but the AP algorithm performs better than the RLS algorithm in terms of the adaptation time, even though their linearity improvements are similar.

4. Conclusion

In this paper, adaptation time was suggested as a possible parameter to evaluate the performance, such as the convergence rate and computational complexity, of adaptive digital filter algorithms. Performance comparisons of an adaptive digital predistorted power amplifier based on the RLS, LMS, and AP algorithms were carried out using the adaptation time. The adaptation speed of the adaptive digital predistorted power amplifier using the AP algorithm was four times faster than those of the amplifiers based on the LMS and RLS algorithms.

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