Adaptive Predistortion Linearization of RF Power Amplifiers Using Lookup Tables Generated from Subsampled Data

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Abstract – This paper discusses a new architecture for adaptive predistortion linearization of power amplifiers, whereby subsampled data (i.e. lower than Nyquist rate) from the input and output baseband waveforms is used to construct a lookup table. The technique utilizes a sampling downconverter that accurately captures a waveform sample in a narrow aperture before digitization. Because aliasing effects are the same for both input and output waveforms, information may be obtained regarding the amplitude and phase distortion characteristics of the power amplifier. The concept is validated using a test bed consisting of two sampling downconverters and a dual channel receiver. Results achieved indicate that 10-12 dB of adjacent channel power ratio (ACPR) improvement may be obtain using sampling rates as low as 33% of the Nyquist rate of the baseband output signal bandwidth.

I. INTRODUCTION

Predistortion linearization has been employed to improve the intermodulation distortion (IMD) introduced by a nonlinear power amplifiers (PAs) [1]. A common architecture uses a lookup table (LUT) stored in digital memory, that adjusts baseband signal amplitude and phase in response to known distortion characteristics. Because PAs distortion characteristics may change in response to temperature, voltage, and aging, adaptive architectures are often employed to correct the LUT values to maintain optimum IMD correction [2]. These employ feedback circuits, which sample input and output signals to calculate the best set of LUT values. It is often assumed that full Nyquist rate sampling must be used to properly reconstruct input and output waveforms to extract all PA distortion characteristics [3]. For multicarrier applications, full rate sampling can become costly, as the bandwidth of the signals often exceed 30 MHz. Moreover, the PA output signal is three to five times larger than the input signal bandwidth due to spectral regrowth created by IMD. Thus for many such applications, sample rates in excess of 300 Msps must be employed if alias-free signal reconstruction is desired. With the required resolution being 12 to 14 bits to obtain sufficient dynamic range, these high sampling rates pose significant challenges to realize cost effective adaptive predistortion systems.

This paper introduces a new predistortion architecture whereby the LUT data may be obtained using sampling rates much lower than full Nyquist rate of the output signal. The concept rests on the fact that the distortion characteristics of the PA may be identified using a set of independent samples, irregardless of where they may have occurred in time, as long as the sample size is adequate to properly map out the AM-AM and AM-PM characteristics of the PA. The key component in this architecture is a downconverter that utilizes a narrow aperture sampling window that effectively aliases the high frequency signal components down into the limited passband of the receiver [4, 5]. In the time domain, this process will preserve the high peaks of the input multicarrier signal envelope, thus enabling the PA to be adequately characterized by comparing the individual pairs of input and output samples.

II. SUBSAMPLING

The subsampling concept used so far is to sample a bandpass signal (that has frequency components only between f_l and f_h) with a sampling frequency below the Nyquist rate [6]. However, to avoid the spectrum overlapping, the sampling frequency f_s must satisfy the limitation as below:

$$\frac{2f_h}{k} \leq f_s \leq \frac{2f_l}{k-1} \tag{1}$$

where k is restricted to integer values that satisfy

$$2 \leq k \leq \frac{f_h}{f_h - f_l} \tag{2}$$

and

$$f_h \leq 2f_l \,. \tag{3}$$

This condition means that f_s must satisfy $2 \cdot BW \leq f_s$, where BW f_h-f_l.

In case of a moderate nonlinear device, however, in this paper we suggest that it can be overlapped for the identification of nonlinearity of a PA whereas the signal cannot be overlapped for the full reconstruction of the original signal. When a complex signal in frequency domain is $S(f)$ at a frequency f_{IF}, and we sample it with f_s, the sampled signal in frequency domain can be expressed with the series of $S(f)$ spaced by f_s, and with the series of

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$S^*(-f)$ spaced by f_s: If we say $S(f-f_{IF})$ as the frequency domain signal at the carrier frequency f_{IF}, the sampled signal is

$$S_{sampled}(f) = \sum_{n=-\infty}^{\infty} S(f-(n\cdot f_s + f_{IF})) + \sum_{n=-\infty}^{\infty} S(-f+(n\cdot f_s - f_{IF})) \quad (4)$$

where n is an integer.

Since we're interested in signals at the lowest frequency for the Analog-to-digital conversion, the signal of interest can be expressed as below:

$$S_{interest}(f) = [S(f-f_{IF}) + S(-f+(f_s - f_{IF}))] \quad (5)$$

The original signal and its image signal starts to overlap when $f_s = BW + 2*f_{IF}$ if we put (5) into a bandpass filter designed to pass from f_{IF} $BW/2$ up to f_s, this is suggested to be used for the nonlinear characterization. Similarly, we can digitize the distorted signal at the output of PA, and then compare both signals in complex domain so that we can identify the AM-AM and AM-PM distortions of the PA. Figure 1 shows the concept of this sampling. In terms of the lowest limit of this subsampling theorem, it is expected to be the sampling limit of the input signal [7].

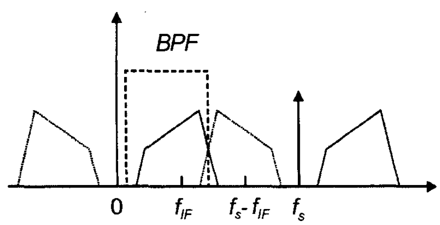

Figure 1: Subsampling concept in Frequency Domain.

This subsampling can be very handy when it comes to a wideband signal because we can still fully identify its characteristics even though a portion of the signal is folded whereas the sampling frequency so far must be at least six times of its bandwidth if we include the 3rd order distortion. Thus we don't need to exactly decide the sampling frequency of a non linear device for its identification needless to say about the mitigation of the system requirements.

III. PREDISTORTION ARCHITECTURE

Since it is not guaranteed to get synchronized between the measured and the generated data, the in-direct learning architecture is adopted for the predistortion as proposed in [8]. The main benefit of this architecture is that it is not required to take the inverse characteristic of a PA therefore, we can predistort a signal without having to restore the original data that may introduce additional characterization error.

Figure 2 shows the architecture of an indirect-learning architecture.

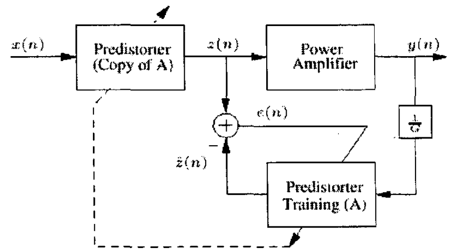

Figure 2: The indirect learning architecture for the predistorter.

IV. LUT ALGORITHM

In order to compensate for the time-varying property of a PA, least-mean-square (LMS) method is used as the adaptive predistortion algorithm:

If we assume the sampled input and output signal of a PA as $z[n]$, and $y[n]$, respectively, the output distorted signal can be expressed as

$$y[n] = G(z[n]) \cdot e^{j\Phi(z[n])} \quad (6)$$

where $G(\cdot)$, and $\Phi(\cdot)$ are the PA amplitude, and phase distortions, respectively. These $z[n]$ and $y[n]$ signals are used to generate a LUT for AM-AM and AM-PM compensation with LMS algorithm as below:

$$LUT(x[n+1]) = LUT(x[n]) + \mu \cdot e[n] \quad (7)$$

where μ is a stability factor, and $e[n] = (z[n]-LUT(y[n]))$.

V. SIMULATIONS

In this section, we illustrate the AM-AM, and AM-PM characterization of a PA with suggested subsampling concept. A polynomial PA model for the Stanford Microwave's SHF-0189 (0.5W) was used for the simulation test signal was 1.2288 MHz IS-95B signal the aperture width for the sample and hold circuit was 10 [nsec]. This PA is believed to have negligible memory effects. Figure 3 and Figure 4 are its characterization results when f_s were 10 MHz, and 5.3 MHz, respectively. The lower sampling frequency was chosen since the input signal spectrum starts to overlap from this value.

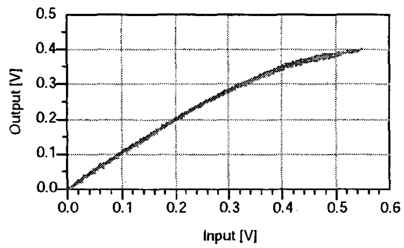

Figure 3: Simulated AM-AM Characteristic of SHF-0189 when f_s 10 MHz.

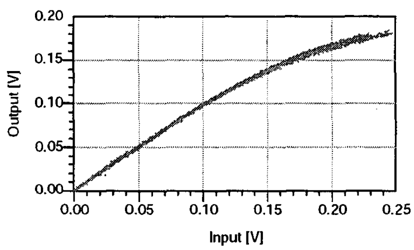

Figure 4: Simulated AM-AM Characteristic of SHF-0189 when f_s 5.3 MHz (67% of output signal is overlapped).

The nonlinear characteristics were clearly identified with f_s 10 MHz of sampling as well as with f_s 5.3 MHz where 67 % of the output spectrum was overlapped with its image signal. The down scale of the latter case is due to the less chance of capturing the signal energy this can be overcome with a proper scale factor for the predistortion.

VI. Measurement Setup

To verify the subsampled predistortion architecture, we constructed a test bed consisting of two sampling downconverters connected to the input and output of a PA, as shown in Figure 5. The outputs of the downconverters were applied to a dual channel receiver and demodulated to calculate the PA distortion. The sampling rates of the downconverters were varied over several times of magnitude, and the receiver bandwidths were adjusted accordingly. We then adapted a LUT to optimally correct the baseband signals, and thus predistort the PA.

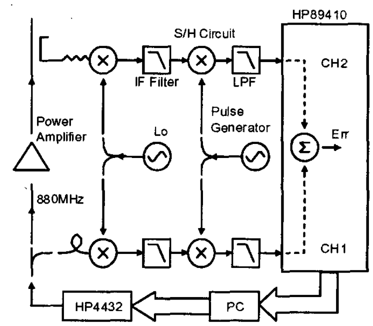

Figure 5: Subsampling Downconverter Test Bed Block Diagram.

The measurement was done with IS-95B signal at 880 MHz. The predistorted PA was 0.5W-Stanford Microwave's SHF-0189. Downconverted IF signals are captured using narrow aperture sample- -hold circuits formed by wideband mixers and a pulse generator. These sampled signals are fed into the Vector-Signal-Analyzer (HP89410) where signals are digitized, and then demodulated into IQ baseband signals. Hence, we can compare amplitudes 'before' and 'after' the PA even though they were aliased due to the low sampling frequency. A PC analyzes this information and generates LUT used to predistort the baseband signals by complex multiplication.

VII. Measurement Results

AM-AM characteristics with f_s 10 MHz and f_s 5.3 MHz were measured. The first case includes all the distortion information within its sampled frequency band, whereas the latter case has aliased 3rd and higher order signals. The lower frequency f_s 5.3 MHz was chosen since that frequency is where the adjacent 'input signal spectrums' start to overlap with each other. This means that 67% of the output signal bandwidths are overlapped at that sampling frequency. The identification results after the narrow-aperture sample and hold circuits are shown in Figure 6, and Figure 7, respectively. From these graphs, we can see that the f_s 10 MHz case shows a clear AM-AM curve, and that f_s 5.3 MHz case does, too although it has a little bit of fuzziness due to some cross talks from the adjacent input signal spectrums

Adjacent-channel power ratios (ACPR) after predistortion are shown in Figure 8 based on these identifications with different sampling rates. They were measured at 885 kHz offset, with 1.2288 MHz IS-95B signals. The number of samples collected at all sample rates was between 1200 and 1600 for each LUT iterations. Several iterations were completed to get the results shown in Figure 8.

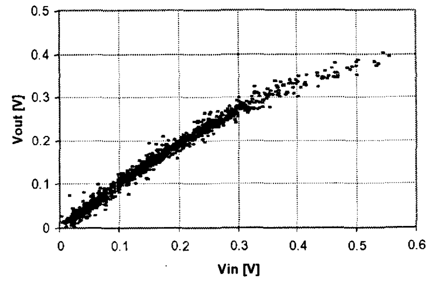

Figure 6: Measured AM-AM Characteristic of SHF-0189 when f_s 10 MHz (includes 3[rd] order distortions).

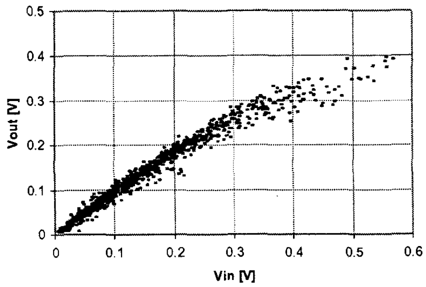

Figure 7: Measured AM-AM Characteristic of SHF-0189 when f_s 5.3 MHz (67% of output spectrum is overlapped).

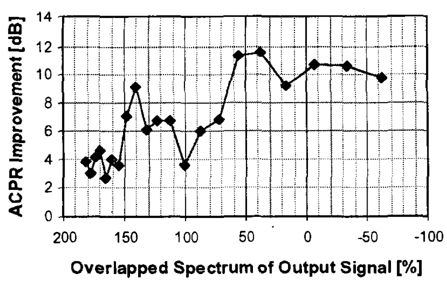

Figure 8: ACPR Improvement Using Subsampling Predistortion Architecture (Negative values in x-axis means that it's above Nyquist Rate.)

The x-axis in this graph indicates the ratio of overlapped spectrum to the output signal spectrum. The negative values in the x-axis mean that spectrums are not overlapped at all the values over 100% mean that the image signal has passed the original spectrum: therefore, it is placed between 0 and f_{IF}. It is clear that the performance of the predistorter was not affected by low sampling rates above 33% of the output Nyquist sampling rate. Below this value, the system identification performance begins to degrade. This phenomenon agrees with the expectations in the previous section.

In this figure, a little bit of bounce in performance around f_s 2 3 MHz (100 150% of overlapping region) is noticeable the reason for this is assumed to be the result of the bandpass sampling of the image signal between 0 and f_{IF} range, although it needs further investigation.

VIII. CONCLUSIONS

A new architecture for LUT generation in adaptive predistortion linearization systems has been developed. Using a narrow aperture sampling downconverter, samples of the envelope were obtained from input and output of the PA. It was seen that sampling below the Nyquist rate of the output signal did not adversely affect the correction capability of the algorithm. Ten to twelve dB of ACPR improvement was obtained for a 0.5W PA at sample rates down to 33% of the Nyquist rate. It was also empirically verified that the indirect learning architecture could be adapted to a predistortion system where the original data fed into the predistorter is not accessible.

For the future work, predistorter with subsampling for a higher power amplifier will be introduced, and the nonlinear identification with memory as well as with wider bandwidth signals will also be included.

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