A NOVEL FREQUENCY SYNTHESIZER CONCEPT FOR WIRELESS COMMUNICATIONS

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Abstract

The design of frequency synthesizers is especially challenging for wireless applications due to the requirements for high spectral purity, high frequency range, and fast tuning together with reasonable power consumption. In this paper, the idea of combining digital and analog techniques for achieving these goals is discussed. The proposed architecture uses I/Q modulation to translate a digitally synthesized tuneable low frequency tone to the final frequency range. In practice, mismatches between the I and Q branch amplitudes and phases as well as the local oscillator leakage can seriously degrade the spectral purity of the synthesized signal. A digital pre-compensation structure is presented to enhance the signal quality. The performance of the proposed synthesizer concept is illustrated using simulation results.

1 INTRODUCTION

In radio communications, sinusoidal signals are used to convert baseband messages into carrier modulated transmission signals and vice versa. Thus, synthesizing high quality oscillator signals is an important part of any radio transceiver. In general, to communicate over any of the possible channels, the oscillating frequency has to be adjustable across the whole system band with tuning resolution equal to channel spacing [1]-[4].

Traditional analog synthesizers are based on multiplying the frequency of a stable reference source using a phase-locked loop (PLL) with a frequency divider in the feedback path [1][2]. In the basic schemes, the output resolution equals the reference frequency and tunability is obtained by using a programmable divider. Digital synthesis techniques, in turn, generate sinusoidal signal samples in digital domain and use digital-to-analog (D/A) conversion to build up the analog waveform. This is usually referred to as direct digital synthesis (DDS) [3]. While the power consumption of digital techniques is generally higher than that of the corresponding analog solutions, low phase noise, high resolution, and high tuning speed have made DDS an appealing alternative in many applications [3][4].

2 SYNTHESIZER ARCHITECTURE

In wireless communications, the demands for fast switching and high (typically GHz) operating frequencies make the design of frequency synthesizers a challenging task [1]-[4]. From the whole terminal point of view, integrability and power consumption are also of great concern. With these aspects in mind, the basic approach here is to generate a low frequency tone digitally, and then mix this signal to the final frequency range. This concept is illustrated at a general level in Figure 1. To avoid radio frequency (RF) filtering, in-phase/quadrature (I/Q) mixing principle is used for sideband suppression. All the tuning can be made in the digital domain and thus a fixed frequency local oscillator (LO) can be used in the mixing stage. Notice that the absolute frequency range of the digital synthesis part can be kept at a relatively low level resulting in feasible demands for D/A converters and reasonable power consumption.

In practical implementations, several non-idealities can distort the spectral purity of the synthesized signal. First of all, due to the finite number of bits in the digital part, some spurious components will appear in the D/A converter outputs, degrading the spectral purity of the actual synthesizer output [3]. Increasing the D/A converter resolution can, however, be used to reduce these spurs to acceptable levels. More challenging problem in this context is the effect of non-idealities in the analog part. Especially in integrated circuit implementations, the amplitudes and phases of the I and Q branches can only be matched with finite relative accuracy [5]. This results in an image tone to appear at the output signal. In addition, the isolation between the mixer LO and output ports is only finite allowing part of the LO signal to leak into the mixer output. However, to enhance the spectral purity of the synthesized signal, digital pre-distortion of the low frequency tone can be applied.

3 SIGNAL ANALYSIS

3.1 I/Q Imbalance, LO Leakage and Pre-Compensation

For analysis purposes, we write the low frequency I and Q signals in continuous-time domain as $I_{NCO}(t) = \cos(\omega_{NCO}t)$ and $Q_{NCO}(t) = \sin(\omega_{NCO}t)$. Furthermore, the I/Q mixer LO signals are written as $I_{LO}(t) = \cos(\omega_{LO}t)$ and $Q_{LO}(t) = g\sin(\omega_{LO}t + \phi)$ where g and ϕ denote the gain and phase imbalances of the mixing stage. With general pre-compensation parameters a_{ij} and $a_{I,Q}$, the I/Q mixer input signals are given by
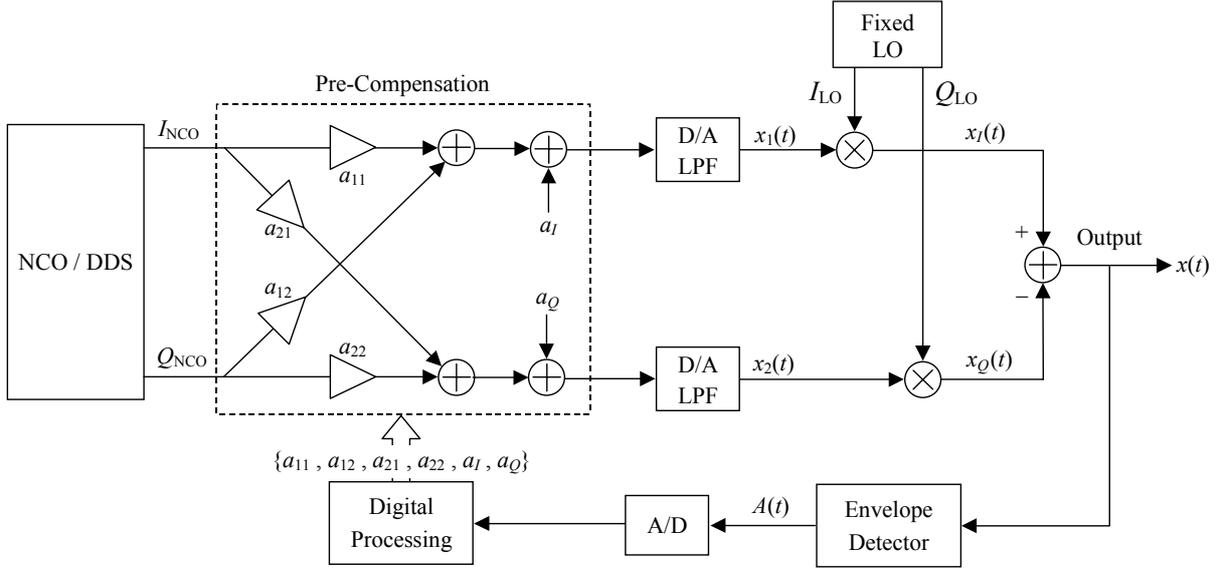
Figure 1: Synthesizer architecture with digital pre-compensation to compensate for the non-idealities of the analog part.

$$x_1(t) = a_{11} \cos(\omega_{\text{NCO}}t + \theta_1) + a_{12} \sin(\omega_{\text{NCO}}t + \theta_1) + a_I$$
$$x_2(t) = a_{21} \cos(\omega_{\text{NCO}}t + \theta_2) + a_{22} \sin(\omega_{\text{NCO}}t + \theta_2) + a_Q$$

(1)

where θ_1 and θ_2 model the relative phase shifts due to the D/A converters and branch filters. Notice that any gain mismatch between the D/A converters and filters can be modeled in the mixer gain imbalance g. Denoting the mixer leakage values by L_I and L_Q, the synthesizer output signal $x(t) = [x_1(t) + L_I]I_{\text{LO}}(t) - [x_2(t) + L_Q]Q_{\text{LO}}(t)$ can be easily shown to be of the form $x(t) = \text{Re}[x_{\text{LP}}(t)\exp(j\omega_{\text{LO}}t)]$ where the lowpass equivalent $x_{\text{LP}}(t)$ is given by

$$x_{\text{LP}}(t) = \alpha e^{j\omega_{\text{NCO}}t} + \beta e^{-j\omega_{\text{NCO}}t} + \gamma .$$

(2)

Thus, the synthesizer output consists of three spectral components: the desired tone at $\omega_{\text{LO}} + \omega_{\text{NCO}}$, the image tone at $\omega_{\text{LO}} - \omega_{\text{NCO}}$ and the carrier leakage tone at ω_{LO} whose relative strengts $|\alpha|^2$, $|\beta|^2$ and $|\gamma|^2$ depend on g, ϕ, $\theta = \theta_2 - \theta_1$, L_I and L_Q as well as on the compensation parameters a_{ij} and $a_{I,Q}$. With practical analog electronics, the image tone and carrier leakage attenuations defined here as

$$L_{\text{IMAGE}} = |\alpha|^2 \big/ |\beta|^2 \quad \text{and} \quad L_{\text{CARRIER}} = |\alpha|^2 \big/ |\gamma|^2$$

(3)

are only around 20…40 dB if no compensation is used. In wireless systems, these levels of synthesizer spurious tones can result in severe interchannel interference (ICI) [4].

Considering the proposed concept, the spectral purity of the synthesizer output can be enhanced with digital pre-compensation. In fact, setting $\beta = \gamma = 0$ (with $|\alpha| \neq 0$) will force the output to be a pure sinusoidal. The corresponding compensation parameters can be easily shown to be of the form

$$a_{11} = 1, \quad a_{12} = \tan(\psi), \quad a_{21} = 0, \quad a_{22} = \frac{1}{g\cos(\psi)}, \quad a_I = -L_I, \quad a_Q = -L_Q$$

(4)

where $\psi = \phi - \theta$ denotes the phase error *difference*.

3.2 Parameter Update Based on Envelope Variations

In practice, the imbalance and leakage levels needed to calculate the compensator parameters in (4) are unknown and need to be measured or estimated somehow. To emphasize implementation simplicity, the approach taken here is to carry out the estimation using only the envelope

$$A(t) = |x_{\text{LP}}(t)| = |\alpha e^{j\omega_{\text{NCO}}t} + \beta e^{-j\omega_{\text{NCO}}t} + \gamma|$$

(5)

of the synthesizer output. As is obvious, the envelope is flat if and only if the image and carrier tones are completely attenuated ($\beta = \gamma = 0$). Instead of estimating g, ψ, L_I and L_Q, a more simple approach is to directly adapt a_{12}, a_{22}, a_I and a_Q to minimize the envelope variation. One possibility is to consider the envelope peak-to-peak (PP) value $d_A = \max\{A(t)\} - \min\{A(t)\}$. Though not strictly parabolic, this d_A-surface having a unique minimum lends itself well to iterative minimization. The true gradient of d_A depends, however, on g, ψ, L_I and L_Q which are unknown. A practical (yet AD-HOC) approach is then simply to adapt only one parameter at a time. The

direction (sign) of the needed one dimensional gradient at each iteration can be determined based on observing the behaviour of d_A between two previous adaptations of the corresponding parameter. The amount of update for each parameter is here formulated as $\lambda_1 d_A / (\lambda_2 + d_A)$ which is bounded in magnitude between zero and λ_1.

4   EXAMPLE SIMULATIONS

To demonstrate the proposed synthesizer concept with digital pre-compensation, some computer simulations are carried out. In the basic simulations, imbalance values of $g = 1.04$, $\phi = 6°$, and $\theta = 1°$ ($\psi = \phi - \theta = 5°$) are used and the carrier leakage levels are $L_I = 0.009$ and $L_Q = 0.004$. Pre-compensation parameters are initialized as $a_{12} = 0$, $a_{22} = 1$, $a_I = 0$, $a_Q = 0$ and are then iteratively updated one-by-one to minimize the peak envelope variation. In general, updating is carried out once per NCO cycle and step-size values of $\lambda_1 = 0.02$ and $\lambda_2 = 0.04$ are used for a_{12} and a_{22}, and $\lambda_1 = 0.002$ and $\lambda_2 = 0.01$ for a_I and a_Q, respectively.

With these example values, the behaviour of the synthesizer output envelope is depicted in Figure 2 verifying successful synthesizer operation. The steady-state operation is reached in 50 iterations or so and the steady-state image and carrier attenuations are in the order of 100...150 dB.

To illustrate the ability to perform the compensation also in time-varying environments, an abrupt change in the non-idealities is tested. The gain mismatch g is changed from 1.05 to 0.98 and the I/Q mixer phase imbalance ϕ from 6° to 3°. Also the LO leakages are changed to $L_I = L_Q = 0.007$. The resulting output envelope is presented in Figure 2 evidencing fast and accurate synthesizer operation also in case of time-varying non-idealities.
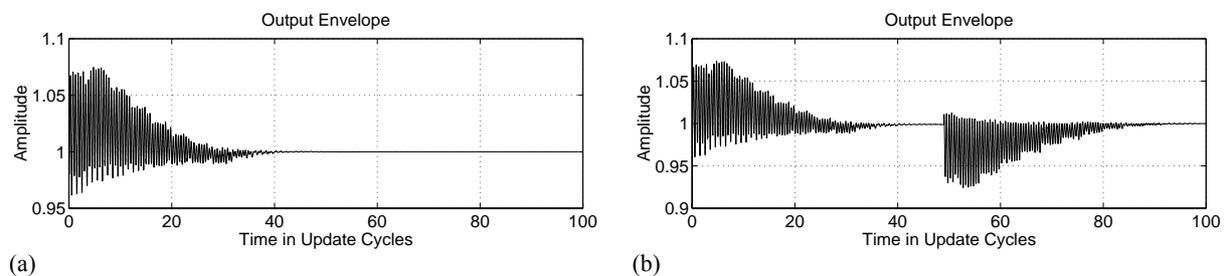
Figure 2: Envelope of the synthesizer output signal during the adaptation. (a): $g = 1.04$, $\phi = 6°$, $\theta = 1°$, $L_I = 0.009$ and $L_Q = 0.004$. (b): the initial values $g = 1.04$, $\phi = 6°$, $\theta = 1°$, $L_I = 0.009$ and $L_Q = 0.004$ are changed rapidly to $g = 0.98$, $\phi = 3°$, $\theta = 1°$, $L_I = 0.007$ and $L_Q = 0.007$.

5   CONCLUSIONS

In this contribution, frequency synthesizer design for wireless applications was considered. To achieve fast switching capabilities with high operating frequencies and reasonable power consumption, the approach was to use I/Q modulation of digitally tunable low frequency tone. The practical I/Q imbalance and LO leakage problems of analog signal processing were then considered and analyzed. Based on this analysis, a digital pre-compensation structure was presented together with a simple yet efficient approach to determine the compensator coefficients. The proper operation of the whole synthesizer was demonstrated using computer simulations both in time-invariant and time-varying situations. Future work should be directed to a detailed evaluation of the finite wordlength effects. Hardware prototyping and/or co-simulation of the analog and digital parts are needed to assess the practical performance of the proposed synthesizer concept.

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