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Adaptive Analog to Digital Converter for Adjacent Channel Interference Rejection

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Abstract: A method for digital encoding of analog signals through adaptive digital predictive encoding is presented. The invented method improves the performance of predictive encoders in the presence of adjacent channel interference. The method involves changing the encoder loop gain in response to the measured bandwidth of the predictive encoder input signal. The input signal bandwidth is measured with a simple digital circuit. Performance results are presented to illustrate the concept and benefits.

1. Introduction

Cost and performance tradeoff of receiver analog to digital converters (ADCs) is a key component of receiver design. In evaluating performance of ADCs dynamic range, sensitivity of characteristics to variations in analog sub-components, and the capability to accommodate wide band signals are typically taken into account. This paper describes an approach to the design of a low cost A/D converter that utilizes these criteria.

Although sampling a signal at the Nyquist frequency retains sufficient information for reconstruction of a band limited signal from its samples, using sampling at the Nyquist frequency requires high cost sharp transition analog anti-aliasing filters. In order to relax anti-aliasing filter requirements over sampling techniques are used.

Oversampling A/D conversion can be accomplished with a conventional "staircase" type, predictive or interpolative architectures. With the conventional "staircase" type quantizer, the accuracy requirements on overall A/D conversion is hard to meet with low cost and simple circuit design. Predictive and interpolative oversampling techniques capitalize on the capability of contemporary integrated circuit technology providing high speed digital circuitry with lower costs relative to precise analog circuitry [Ref. 3]. These techniques obtain accuracy through usage of coarse, but, very fast circuitry and suitable digital signal processing to improve the effective accuracy of conversion. Digital circuits also do not suffer from variations due to circuit manufacturing tolerances, aging, and temperature that effect the performance of analog circuits.

Predictive and interpolative encoders use feedback techniques to digitally reconstruct the analog input signal. In predictive encoders the error between the input signal and local digital estimate of the input signal is quantized with an analog to digital converter and then filtered, on the other hand in interpolative coders, the error is first filtered with a discrete time filter and then A/D converted before being fed back [2]. This re-ordering of filtering and A/D conversion in the feedback path leads to significantly different operational characteristics and brings different levels of sensitivity to analog component variations. The feedback filter in a predictive encoder can be implemented with a digital filter eliminating dependence on analog circuit imperfections. On the other hand, interpolative coding A/D conversion systems, by their nature have to have analog components feedback filters. Thus, despite several advantages of interpolative coding, requirements on the analog components present implementation challenges.

In wireless communication systems presence of adjacent channel interference (ACI) signals produce a significant receiver design challenge. ACI signals are at close spectral proximity and they can be significantly stronger than the desired signal. ACI can be removed either with an analog filter before the A/D conversion, or with a digital filter after the A/D conversion. There are several advantages to digital filtering such as lower cost, independence of manufacturing tolerances, effects of aging, and temperature. However, for digital filtering option, dynamic range requirements on the ADC may become stringent with strong ACI signals as the ADC should provide a digital representation of the analog signal composed of the desired signal and the ACI with sufficiently low distortion in the desired signal bandwidth.

The requirement of digitizing a large bandwidth signal in the presence of ACI offers challenges in predictive encoder design. There is a limit to the rate of change of a signal out of a predictive encoder [Ref. 2]. The maximum rate of change can be defined as the maximum instantaneous derivative of a signal. If a signal with a higher rate of change than this maximum rate is applied to the predictive encoder, the encoder cannot follow the rapid variations of the input signal (slope overload) and A/D conversion quality is significantly reduced. Typically the maximum rate of change of a signal increases signal bandwdith increases. ACI exposing a mobile communication receiver increases the bandwidth of the signal that is to be quantized. Hence a properly designed ADC should be capable of converting wide band signals when exposed by ACI without slope overload. In order to design such an ADC choosing the system parameters so as to accommodate the signals that have the widest bandwidth (worst case signal scenario) without slope overload is a possible design approach. This can be achieved by increasing the sampling frequency and update frequency of the predictive encoder feedback loop. However, this is not desirable as circuits that operate at higer frequencies typically cost more and consume higher power. In the alternative design approach presented in this paper we utilize the fact that ACI signals are not always present, and, if they are present, they may not be full strength. Instead of designing for the largest input signal bandwidth, the ADC is designed to adapt its parameters in response to changes in input signal bandwidth.

The maximum signal rate of change that a predictive encoder can accommodate is connected to the choice of the open loop gain of the predictive encoder loop and this maximum rate can be increased by increasing the loop gain. However, increased loop gain leads reduced resolution for the ADC output signal. The distortion that is introduced to due to finite resolution of representation of the analog signal with discrete levels is called the granular noise. As the loop gain is increases, the granular noise increases. Thus, the choice of loop gain involves a trade off between avoding slope overload and reducing granular noise level.

This paper introduces an adaptation method for adapting predictive encoder parameters in response to changes in the bandwidth of the input signal and trading off slope overload and granular noise performances.

2. System Definition

In communications receivers design it is desirable to perform the A/D conversion as early as possible in the receiver chain. Thus reducing the amount of analog circuitry while increasing the digital circuitry. As mentioned earlier, digital circuitry typically cost lower and does not display variations between individual one compenet of circuit to another, and the system charactersitics does not change with aging and temperature. A popular receiver architecture is the low IF architecture where A/D conversion at an IF frequency is followed by digital down conversion. Digital down conversion is not plagued by usual problems of gain and phase balancing issues of analog down conversion. Although the design principles of this paper applicable to other reciver architectures including superheterodyne and direct down conversion, in this paper the presentation is for a low IF type receiver chain with a bandpass predictive encoder used for A/D conversion.

Consider the Frequency Adaptive Predictive Encoder block diagram shown in Figure 1 that includes a traditional predictive encoder and an Adaptive Parameter Control Unit (APCU). This paper describes the design of APCU and describes its utilization in the predictive encoder.

The predictive encoder includes a one bit ADC, a digital predictive filter and a DAC. The encoder constructs a digital copy of the analog input signal and a prediction error signal is generated at the adder output. This error signal is digitally coded by the 1-bit ADC. The output of the 1-bit ADC is then used as an input to the predictive filter to produce the next sample of the predicted digital copy of the analog input signal.

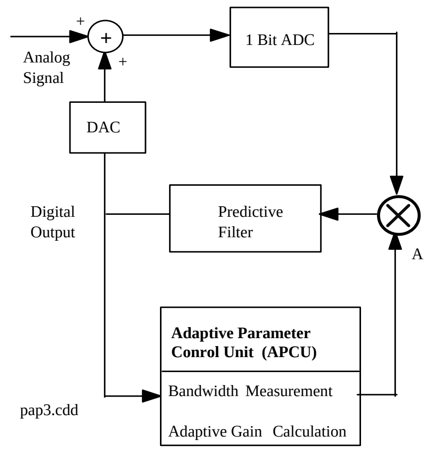

Figure 1 Frequency Adaptive Predictive Encoder

A convenient bandpass implementation of the predictive encoder, the system sampling frequency, f_s is chosen to be four times the bandpass signal carrier frequency, f_c. This yields simple implementation of predictive sub-filter stages and eases implementation of down conversion following the A/D conversion. For this particular choice of the sampling and carrier frequencies, the error signal between the input signal and locally generated copy of the signal is shown to be obtained by addition rather than subtraction.

Figure 2 shows the block diagram of a two stage predictive filter with weighting factors G_1 and G_2 for the sub-filter stages. As there are two stages in the feedback loop, the encoder is said to be of 2^{nd} order. The technique described in this paper is applicable to encoders of arbitrary order.

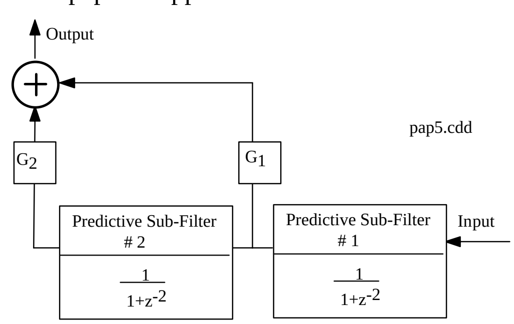

Figure 2 Block diagram of predictive filter.

The APCU generates a suitable value of the loop gain scaling factor "A" in accordance with APCU input signal bandwidth. The APCU input signal is a predicted digital copy of the analog input signal. The APCU is constructed with digital circuitry. Consider the block diagram of the APCU shown in Figure 3. The input signal is passed through an ALC block to keep the signal at a constant level. APCU measure the bandwidth of the signal at its input rather than responding to signal level changes. The signal is then passed through a bandpass filter centered at the output carrier frequency. The input and the output of a bandpass filter are inputs to two nonlinear detectors each of which includes a memoryless nonlinearity and low pass filter (LPF). In the implementation shown, average of the absolute values of these two signals are measured. Power detectors or root mean square (rms) type detectors can also be used.

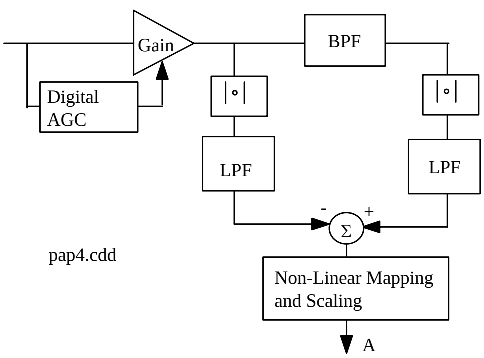

Figure 3 Adaptive Parameter Control Unit (APCU).

For a signal that has a wider frequncy arounf f_c, more significant portion of the signal will be filtered out by the BPF and the signal level difference between the LPF outputs measured by the subtracter at the LPF ouputs will be larger. Thus, the subtractor output is a metric that is proportional to the bandwidth of the signal at APCU input. This metric is mapped by a an appropriate scaling function to appropriate values of predictive encoder loop gain.

3. Simulation Results

In order to illustrate the performance of frequency adaptive predictive encoding and its performance compared to non-adaptive predictive encoding we have conducted computer simulations of the predictive encoder with and without the adaptation. As goal of this study is to illustrate the technique rather than provide performance figures for a particular communications system, we have identified system performance by using sinusoidal signals. As mentioned earlier, we have focused on low IF bandpass A/D converter application with IF frequency of 6.5 MHz and sampling frequency of 26 MHz.

Figure [4] shows Signal Power to Distortion Power Ratio (SDR) versus encoder input root mean square (r.m.s.) signal level relative to saturation amplitide level for the feedback DAC for non-adaptive predictive encoder. The results shown for five different signal frequency offset values relative to IF frequency: Δf = 100 KHz, 200 KHz, 300 KHz, 400 KHz, 500 KHz. SDR is defined as the ratio of the signal power relative to total distortion power at the encoder output (wideband measurement of distortion rather than measurement in a limited bandwidth is utilized). The SDR in each case increases as the input signal level increases (for higher input signal levels, the signal power is higher relative to constant granular noise of the digital representation of the signal by the predictive encoder). The SDR for Δf = 100 KHz, drastically deteriorates at input r.m.s. signal amplitude of 0 dB. The input signal values reach beyond the dynamic range of the feedback DAC of the predictive encoder loop for this input amplitude and the system can no longer track the input signal. For Δf = 200 KHz, 300 KHz, 400 KHz and 500 KHz, output SDR deteriorates drastically at input signal levels that decrease as the input frequency increases. This is due to slope overload - the predictive encoder can only track signals with a certain level of rate of change. As the input signal frequency is increased, the maximum signal amplitue that can be tracked has to be reduced for the loop to be able to track the signal as both the amplitude and the frequency of a sinusoid contributes to the signal rate of change.

The input signal level to the receiver ADC is typically kept at a constant pre-set input r.m.s. signal level with an Automatic Gain Control (AGC) circuit. Figure [4] shows that, in order to accommodate signals with higher frequency offset relative to the IF frequency and avoid slope overload, the AGC setting should be reduced significantly. AGC setting level of –23 dB below DAC saturation point is required to ascertain conversion of signal with Δf = 500 KHz. As a result, the maximum SDR that the non-adaptive predictive encoder will provide is limited to about 15 dB.

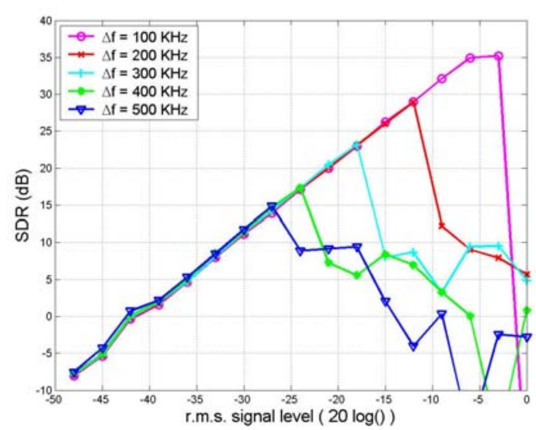

Figure 4 SDR vs. Input r.m.s. signal level for non- adaptive predictive encoder.

In Figure [5] we show SDR versus the (r.m.s.) input signal level for the adaptive predictive encoder. In comparison to Figure [4] it can be seen that the dynamic range of the converter for signal with high Δf values are significantly increased. The predictive encoder is operational without saturation at input r.m.s. level of -6 dB for Δf = 500 KHz. As the system is adapted by changing its feedback loop gain in response to signal bandwidth, the system has different operation curves for signals with different frequency offsets. As Δf increases, the loop gain increases which results in a coarser approximation of the input signal and lower SDR. Thus, by compromising the output SDR, slope overload is avoided. This is the main trade off of adaptive design. The trajectory for each Δf is determined by the adaptive gain profile obtained from the APCU block for the particular Δf.

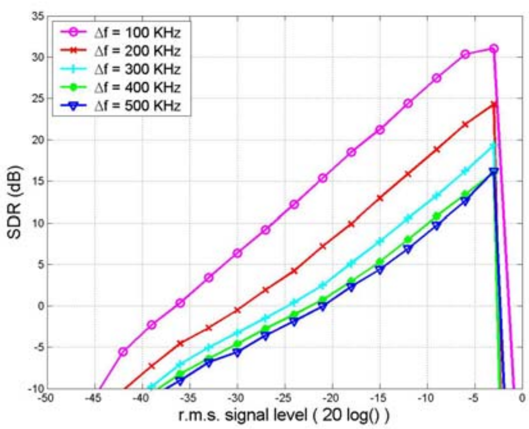

Figure 5 SDR vs. Input r.m.s. signal level for adaptive predictive encoder. .

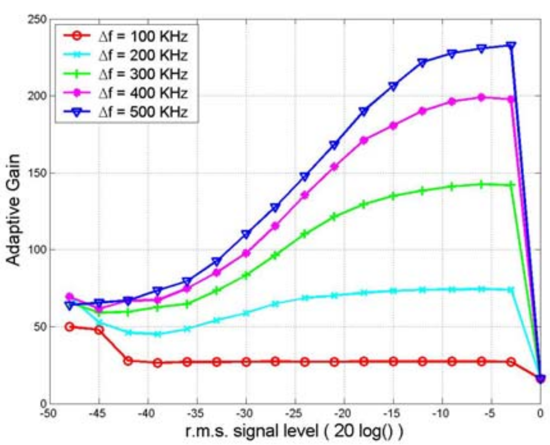

Figure 6 Adaptive gain vs. Input r.m.s. signal level.

In Figure [6] the adaptive gain at APCU output that leads to performance curves shown in Figure [5] are shown. Adaptive gain is plotted for the same input signal r.m.s. range and for the same frequency offsets. As is expected, the the adaptive gain increases as Δf increases.

4. Conclusions

We have introduced adaptive predictive digital encoding as an attractive solution for A/D conversion in wireless mobile communications receivers as by utilizing a digital predictive filter in the predictive encoder feedback loop, dependence on analog circuit imperfections such as circuit manufacturing tolerances, aging, and, temperature are eliminated. We have also shown that performance of predictive encoding can be improved with frequency adaptive encoding. With the proposed adaptation method, the effects of slope overload can be eliminated for high frequency signals.

A particular use of the technique described is in systems where sensitivity performance is of importance. As the signal at sensitivity levels is already corrupted by high level of noise, it is desirable the ADC contribute very little distortion. Thus, high SDR is required for the sensitivity operation. On the other hand, the test conditions in the presence of ACI are such that there is very little noise effecting the system Thus the receiver may survive with a given lower SDR in the presence of ACI provided tboth the ACI and the desired signall is digitized satisfactorily with the encoder.

5. References

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