# Adaptive algorithm for increasing image rejection ratio in low-IF receivers

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An adaptive method for compensating mismatch effect on an I/Q demodulation is presented. It is based on the correlation between the desired and image band that appears when there is a mismatch in any branch. The proposed algorithm is specially intended for narrow-band low-IF receivers. No calibration source is neccesary. Up to 91 dB of image rejection ratio is achieved with narrow-bandwidth signals (up to 6.25% of the sampling frequency).

*Introduction:* Ideal I/Q demodulators would achieve an infinite image rejection ratio (IRR). However, mismatches between their branches produce a leak of the desired signal which appears on the image band, and vice versa. Hence, IRR decreases. Some compensation algorithms use a calibration source , however the proposed method takes advantage of the leaked signals to avoid including additional hardware. Since the leaked signal, that appears in one band, belongs to the opposite band, both bands are partly correlated. Thus, a least mean square (LMS) algorithm can be used to make zero the correlation between bands, and therefore to eliminate both leaked signals, which will improve the IRR. The step-size factor in the LMS algorithm has been adaptively programmed. Then, at the begining, the step size is large, so the algorithm converges fast but with a low resolution to the optimum values, and at the end, the step size is small to achieve an accurate resolution of the compensation factors.

*Adaptive method:* The proposed method is illustrated in Fig. 1. The RF signal is IQ demodulated by means of imbalance mixers, and digitalised by A/D converters. The mismatch compensation block is the first stage of the algorithm. By developing the compensation matrix equations presented , two channel compensation equations are obtained:

$$\begin{aligned} Output\_CHI &= Input\_CHI \times \beta \times (1+\alpha) \\ &\quad + Input\_CHQ \times (1-\alpha) \end{aligned} \tag{1}$$

$$\begin{aligned} Output\_CHQ &= Input\_CHI \times \beta \times (1-\alpha) \\ &\quad + Input\_CHQ \times (1+\alpha) \end{aligned} \tag{2}$$

where $\alpha$ and $\beta$ are the compensation factors of the proposed adaptive algorithm, which will be described later on in this Letter.

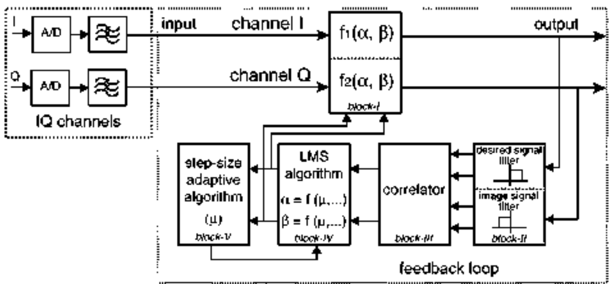


**Fig. 1** *Block scheme of adaptive mismatch compensator*

Equations (1) and (2) calculate the output level of the system. The next blocks are located in a feedback loop, which iterativily calculates the optimum $\alpha$ and $\beta$ values. Block-II and block-III are based on the method presented . Block-II splits up the complex spectrum in positive and negative frequencies (desired band and image band). This block would be ideally accomplished by two complex filters. However, since there is no interest in the phase information of the signal, it is computationally faster and simpler to calculate the fast Fourier transform (FFT) and writing zeros in the negative spectrum of the proccesed signal in order to get the desired band, and writing zeros in the positive spectrum to get the image band. Block-III calculates the normalised correlation between the two bands which have been split up by block-II. The aim of block-IV is to apply an LMS algorithm to the correlation calculated by block-III in order to reach the suitable $\alpha$ and $\beta$ factors which will make zero the correlation between image and desired band. This idea is based on the LMS algorithm proposed . Although the LMS algorithm looks for the gradient that compensates the error power between the two channels, the aim of the algorithm proposed in this Letter is to eliminate the correlation between the image band and the desired band without an external calibration source. The two equations, that obtain the appropiate $\alpha$ and $\beta$ values in each iteration, are:

$$\alpha(t+1) = \alpha(t-1) + 2 \times \mu \times Imag(correlation) \tag{3}$$

$$\beta(t+1) = \beta(t-1) - 2 \times \mu \times Real(correlation) \tag{4}$$

where $\mu$ is the step-size of the LMS algorithm. The previous calculated correlation values are complex numbers, so *Real(correlation)* and *Imag (correlation)* define the real and image part of that number. $\alpha$ and $\beta$ values will be used in the subsequent iterations.

Block-V defines the speed of convergence and the resolution of $\alpha$ and $\beta$ values. The latter features are given by the step-size factor $\mu(0 < \mu < 1)$ of the LMS algorithm. Initially, a large value is assigned to $\mu$. Thus, although the compensation algorithm fastly reaches aproximated values to the optimum ones ($\alpha$, $\beta$), the resolution of this aproximation is low and therefore a relatively high error is appreciated with regard to the optimum values. The reduction of this error is achieved by decreasing $\mu$, which reduces the convergence speed. Consequently, $\alpha$ and $\beta$ converge fast to values close to their optimum ones. Few iterations later, a small value of $\mu$ achieves very accurate $\alpha$ and $\beta$ factors.

When this algorithm is applied to a fixed point digital signal processor (DSP), the numeric resolution of the device is important, because high IRR receivers/demodulators (>50 dB) have small values of correlation between the image and desired band. Hence, the numeric values, that are taken by the adaptive algorithm presented, cannot calculate accurate compensation factors. An easy trick to solve this problem is based on the generation of an internal mismatch, which is taken into account during every iteration. This additional mismatch will increase the numeric values of the correlation. Thus the dynamic range of the DSP registers are better used, since the registers work with more bits.

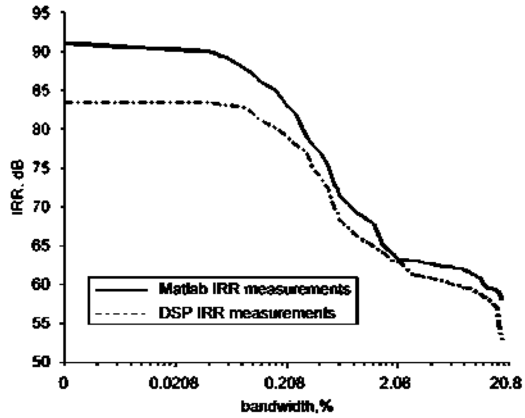


**Fig. 2** *IRR against Percentage bandwidth*

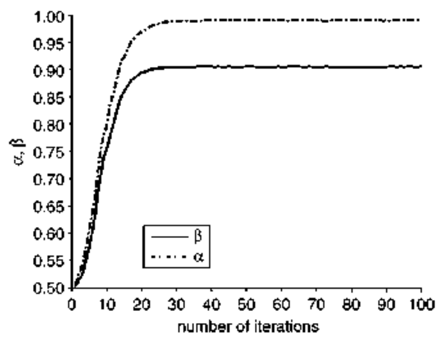


**Fig. 3** *Convergence time*

*Results:* Simulations and real-time measurements were taken on an I/Q demodulator prototype. First, the algorithm was tested in Matlab. Then, a test-bench made up by a low-IF I/Q demodulator directly connected to an Analog Devices ADSP BF533 board was used to check the algorithm. The test signal used was filtered Gaussian noise. Blocks of 512 I/Q samples were processed each iteration. Fig. 2 illustrates the IRR achieved against the bandwidth of the signal. When the bandwidth of the input signal was narrow, the image signal was rejected 91 dB in the Matlab simulation and 83.4 dB in the test-bench. The algorithm achieved more than 60 dB of IRR in a 6.25% bandwidth with regard to the sampling frequency. Convergence measurements were taken for filtered Gaussian noise of 1.25% bandwidth. Fig. 3 illustrates the convergence time of $\alpha$ and $\beta$ factors with an initial value of 0.5. In this case, we can approximately consider that from the 26th iteration the algorithm converges.

*Conclusion:* The proposed adaptive algorithm efficiently achieves high values of IRR (up to 91/83.4 dB) with low computational cost for narrowband signals. No calibration source is neccesary, which reduces the hardware system complexity. Moreover, convergence time is short, therefore any change in I/Q demodulator performance is quickly corrected by the algorithm. As a result, the algorithm maintains almost a constant value of IRR during the demodulator's operative life.

## References


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