I/Q MISMATCH COMPENSATION IN ZERO-IF OFDM RECEIVERS WITH APPLICATION TO DAB

Alexander R. Wright, Patrick A. Naylor

Department of Electrical and Electronic Engineering, Imperial College, London, UK

ABSTRACT

This work addresses the I/Q mismatch problem which arises due to analog component tolerances in zero-IF OFDM receivers. The application of this work is to Digital Audio Broadcasting (DAB). The approach is to employ a decision directed LMS-based frequency-adaptive equalizer, applied to OFDM demodulated carriers. The equalizer is trained initially using the phase reference symbols in each frame and is switched to a decision directed mode when the error reaches a defined level. In the decision directed mode, the output is taken prior to the threshold decision device in order to present to the subsequent Viterbi decoder a signal suitable for soft decisions. Simulation results demonstrate convergence within six frames; the particular convergence rate depending on choice of step-size and number of carriers per group employed. Even with large phase and amplitude imbalances, 25° and 5dB respectively, in a DAB system the compensation algorithm attains a reduction in raw BER from $10^{-1.5}$ to 10^{-5} with a 15dB SNR.

1. INTRODUCTION

Zero-IF and low-IF architectures make use of quadrature demodulation techniques which theoretically provide infinite image rejection in the down converted result. However, in practice they will be subject to mismatches between the in-phase (I) and quadrature (Q) branches of the receiver. This is known as I/Q mismatch and is introduced primarily at two different points; in the local oscillator (LO) components and in the subsequent mixers and filters of the receiver.

The mismatch introduced is a result of the finite tolerances of the analog components. The LO introduces an imbalance due to the impracticality of producing a perfect 90° phase shift. Mismatch of the mixers and filters may also introduce imbalances.

Using relatively good quality analog components one can obtain a phase imbalance of <2° and an amplitude imbalance of <2% are achievable, resulting in a 30-40dB image attenuation in an IF architecture [1]. However, cost of such analog components is high. An alternative strategy is to employ cheaper analog elements and add subsequent compensation. This latter approach is of great interest practically and forms the basis of the motivation for this work.

2. I/Q MISMATCH IN OFDM RECEIVERS

Figure 1 is a simplified DAB zero-IF receiver that is used to model the I/Q mismatch in π/4-DQPSK OFDM broadcasting system. It comprises a quadrature RF demodulator, an OFDM demodulator and a π/4-DQPSK decoder.

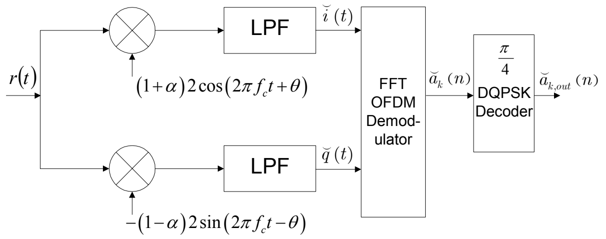

Figure 1: Zero-IF architecture of DAB OFDM receiver.

The received RF signal $r(t)$, at carrier frequency f_c, is defined as:

$$r(t) = [b(t) + m(t)]e^{j2\pi f_c t}, \qquad (2.1)$$

where $m(t)$ is the complex envelope of the additive noise and $b(t)$ is the transmitted OFDM signal. $\breve{b}(t)$, the received OFDM signal, is defined as [2],

$$\begin{aligned}\breve{b}(t) &= \breve{i}(t) + j\breve{q}(t) \\ &= \frac{1}{L}\sum_{k=-\frac{L}{2}}^{\frac{L}{2}-1} \breve{a}_k(n)\, e^{j2\pi k t f_s}, \end{aligned} \qquad (2.2)$$

where L is the total number of carriers, with a channel spacing between two adjacent carriers of f_s, $\breve{i}(t)$ and $\breve{q}(t)$ are the I and Q branches of the zero-IF received signal, $\breve{a}_k(n)$ is the received complex symbol of the k^{th}

carrier at sample index n and $\breve{a}_{k,out}(n)$, in Figure 1 is the symbol outputted from the π/4-DQPSK decoder.

The phase imbalance is defined as 2θ in degrees and the amplitude imbalance is β in dB and is defined using [2]:

$$\alpha = \frac{10^{\beta/20} - 1}{10^{\beta/20} + 1}. \tag{2.3}$$

From the analysis of the quadrature receiver and the OFDM demodulator an expression for the received complex symbol is obtained [2]:

$$\begin{aligned} \breve{a}_k(n) &= \left[a_k(n) + \alpha a_{-k}^*(n)\right]\cos(\theta) \\ &\quad -j\left[\alpha a_k(n) - a_{-k}^*(n)\right]\sin(\theta), \end{aligned} \tag{2.4}$$

where $a_k(n)$ is the transmitted complex symbol, $a_{-k}(n)$ is the transmitted symmetric carrier symbol and * represents complex conjugation.

Figure 2 shows the 'ideal' received constellation (o) with the 'non-ideal' results of the I/Q mismatched received constellation (x), corrupted with 10° phase and 2dB amplitude imbalance. Therefore the 'non-ideal' points are due to the interference from the symmetric carrier, which can be one of four different values for each value of the received carrier of interest.

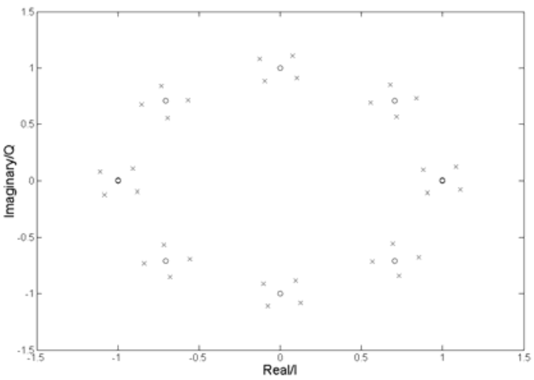

Figure 2: The effect of an I/Q mismatch of 10° phase and 2dB amplitude imbalance on the received symbols.

3. ADAPTIVE EQUALIZATION

Previous work [3] has proposed a compensation technique for I/Q mismatch in Digital Video Broadcasting (DVB) systems for which pilot carriers are always available [4]. In DAB systems such pilot carriers are not available but instead, phase reference symbols are transmitted once per frame. Our approach to the problem in DAB is to develop an adaptive equalizer that can operate without the pilot carriers. This leads to a system which is initially trained on the phase reference symbols then switches to a decision directed mode of operation.

Using Equation (2.4) and a similar expression for the conjugate symmetric carrier, an expression for the transmitted complex symbol can be formed for the k^{th} carrier of interest:

$$\begin{aligned} a_k(n) &= \left\{\frac{\cos(\theta) - j\alpha\sin(\theta)}{\cos(2\theta)\left[1-\alpha^2\right]}\right\}\breve{a}_k(n) \\ &\quad - \left\{\frac{\alpha\cos(\theta) - j\sin(\theta)}{\cos(2\theta)\left[1-\alpha^2\right]}\right\}\breve{a}_{-k}^*(n). \end{aligned} \tag{3.1}$$

Equation (3.1) demonstrates that a structure comprising two single-tap complex transversal filters is sufficient to correct for the I/Q mismatch that has been introduced. Such a structure is presented in Figure 3, where $\breve{a}_k(n)$ is the complex received symbol for the k^{th} carrier, $\breve{a}_{-k}^*(n)$ is the received symmetric carrier symbol, $y(n)$ is the estimate of the transmitted symbol and $d(n)$ is the reference symbol for the adaptive system.

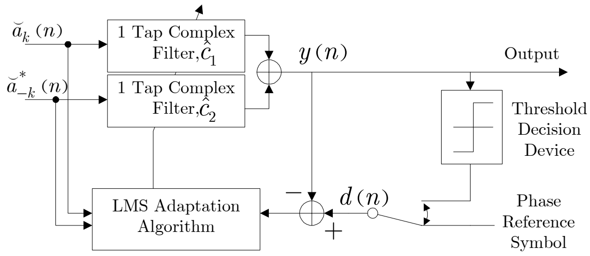

Figure 3: Modified adaptive decision directed equalizer.

It should be noted that in Figure 3, the output of the decision directed equalizer is taken before the threshold decision device rather than after as would be a more common choice [5]. This is done so as to present the subsequent Viterbi decoder a signal suitable for 'soft' decisions.

The reference signal, $d(n)$, is either the phase reference symbol in the data aided training phase or the output of the threshold decision device in the decision directed mode of the adaptive process.

The LMS algorithm is used with this compensation technique due to the low complexity of the algorithm, making it suitable for real time systems and practical for integration into receiver designs. However, there is a problem in that the phase reference symbols are constant for each carrier [6], thus resulting in the autocorrelation matrix of the input data being singular. Hence the LMS algorithm is unable to converge to the optimum coefficients. The approach taken here is to use the data from groups of size $M \geq 2$ adjacent carriers to adapt the adaptive algorithm for each symbol, so that convergence can be obtained. The resulting algorithm is shown in Table 1.

Parameters	L = Total no. of carriers in each frame. M = Number of carriers in each group. μ = Step-size parameter. $\hat{\mathbf{c}}(n) = \left[\hat{c}_1(n) \quad \hat{c}_2(n)\right]^T$
Initialize	$\hat{\mathbf{c}}_0(0) = \left[0 \quad 0\right]^T$
Algorithm	for $p = 0,1,2,\ldots,\dfrac{L}{M}$ for $i = 0,1,2\ldots,(M-1)$ $\breve{\mathbf{a}}_i(n) = \left[\breve{a}_{(pM+1+i)}(n) \quad \breve{a}^*_{-(pM+1-i)}(n)\right]^T$ $y_i(n) = \hat{\mathbf{c}}_i^H(n)\breve{\mathbf{a}}_i(n)$ $e_i(n) = d_i(n) - y_i(n)$ $\hat{\mathbf{c}}_{i+1}(n) = \hat{\mathbf{c}}_i(n) + \mu\breve{\mathbf{a}}_i(n)e_i^*(n)$ $\hat{\mathbf{c}}_0(n+1) = \hat{\mathbf{c}}_M(n)$ end end

Table 1: The Equalizer Algorithm.

Using a single adaptive filter to compensate for $M \geq 2$ adjacent carriers not only enables correct convergence of the LMS algorithm but also provides an implementational saving. However, this method makes the assumption that the I/Q mismatch of the system varies negligibly over the bandwidth of the M carriers in a group. This is an important consideration and in our simulations $M = 2$ has been used which gives the finest frequency resolution.

4. SIMULATION RESULTS

The effectiveness of the compensation scheme that has been outlined is demonstrated in the following DAB simulations. The results are in two separate sets. The first set of results shows constellation diagrams of the received symbols for fixed phase and amplitude imbalance with an ideal channel. The second set of results shows plots of raw bit error rate (BER) surfaces for varying phase and amplitude imbalance with a fixed AWGN channel at SNR of 15dB.

4.1.Constellation Diagrams

The data has been corrupted with an I/Q mismatch of 10° phase and 2dB amplitude imbalance using an ideal channel. Figure 4 is the constellation diagram of the corrupted uncompensated received symbols and Figure 5 is the constellation diagram of the compensated received symbols after five frames in the training mode and one frame in the decision directed mode of the adaptive algorithm. It can be seen that the equalizer has correctly compensated for the I/Q mismatch and results in the 'ideal' constellation (o) as shown in Figure 2. At convergence the weight error vector norm is better than 10^{-5}.

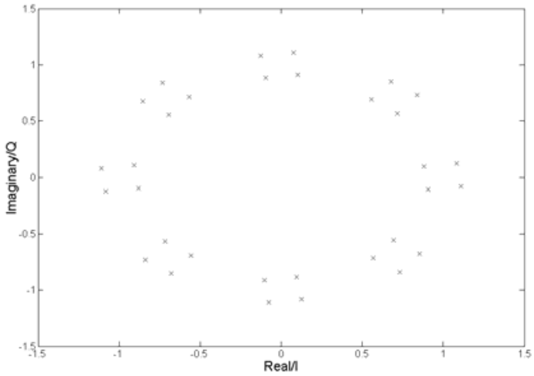

Figure 4: Uncompensated received symbols corrupted with 10° phase and 2dB amplitude imbalance.

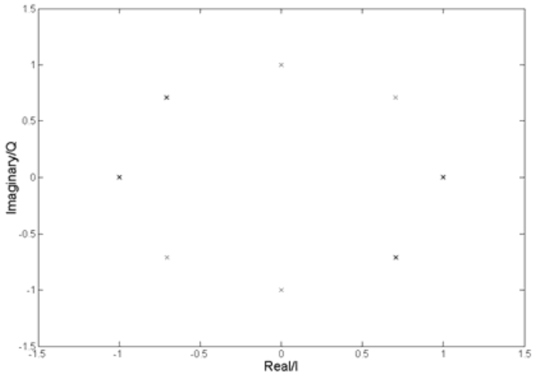

Figure 5: Compensated received symbols corrupted with 10° phase and 2dB amplitude imbalance after six frames of data.

4.2.Raw BER Curves

The raw BER results have been obtained using an additive white Gaussian noise channel with a SNR of 15dB. Raw BER surfaces have been plotted to demonstrate the effectiveness of the algorithm over a wide selection of values for phase and amplitude imbalance. Figure 6 demonstrates the effect that varying the phase and amplitude imbalance for a given SNR has on the raw BER. Figure 7 demonstrates effectiveness of the compensation algorithm using the same conditions as those in Figure 6 and the results taken after 153 iterations of decision directed mode.

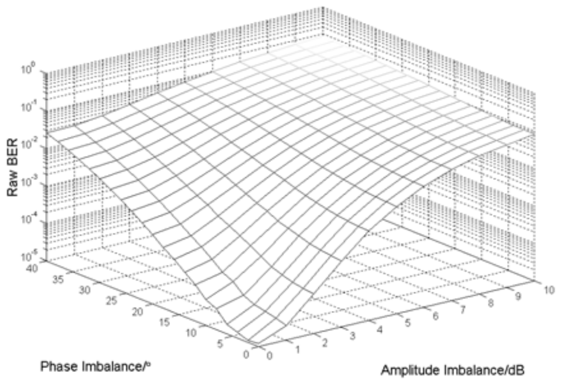

Figure 6: Uncompensated data with 15dB SNR AWGN, with varying phase and amplitude imbalance.

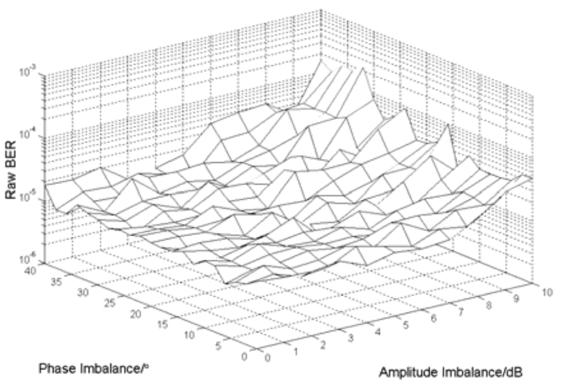

Figure 7: Compensated data with 15dB SNR AWGN, with varying phase and amplitude imbalance.

5. CONCLUSION

The work that has been presented describes a new approach for compensating for I/Q mismatch in an OFDM DAB receiver. An LMS adaptive equalizer has been employed to provide an output suitable for 'soft' decision decoding in a Viterbi decoder. The simulation results for the constellation diagrams and raw BER surface plots both demonstrate effective compensation for I/Q mismatch. For example, even with large phase and amplitude imbalances, 25° and 5dB respectively, in a DAB system the compensation algorithm attains a reduction in raw BER from $10^{-1.5}$ to 10^{-5} with a SNR 15dB . The low complexity of the algorithm makes it a suitable option for real-time operation as well as practical for integration into receiver designs.

6. REFERENCES

1. Valkama M, Renfors M, and Koivunen V, "Advanced Methods for I/Q Imbalance Compensation in Communication Receivers", *IEEE Transactions on Signal Processing*, vol. 49, no. 11, pp.2335-2344, Oct. 2001.

2. Chia-Liang Liu, "Impacts of I/Q Imbalance on QPSK-OFDM-QAM Detection", *IEEE Transactions on Consumer Electronics*, vol. 44, no. 3, pp.984-989, Aug. 1998.

3. A.Schuchert, R.Hasholzner, and P.Antoine, "A Novel IQ Imbalance Compensation Scheme for the Reception of OFDM Signals", *IEEE Transactions on Consumer Electronics*, vol. 47, no. 3, pp.313-318, Aug. 2001.

4. "Digital Broadcasting Systems for Television, Sound and Data Services: Framing Structure, Channel Coding and Modulation for Digital Terrestrial Television.", EBU/ETSI JTC-ETS 300421, 2002.

5. Haykin S, *Adaptive Filter Theory*, Prentice Hall, 1996

6. "Digital Audio Broadcasting (DAB) to mobile, portable and fixed receivers", EN 300-401, pp.1-223, Apr. 2000.