# Analog Front-End Cost Reduction for Multi-Antenna Transmitter

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*Abstract – Multi-antenna systems provide the option to enhance the data rates and to improve the overall system performance. Therefore it is important that the transmitter provides high signal accuracy for all different signal branches. On the one hand high signal accuracy can be reached by employing expensive, high-end analog front-ends. On the other hand a more cost efficient solution might be a low cost analog front-end in combination with digital, software-based pre-adjustment algorithms to guarantee high precision output signals. A direct-conversion transmitter analog front-end architecture for an OFDM multi-antenna system will be proposed and analog filter imperfections are compensated based on a digital filter pre-equalizer. Because of the direct-conversion technique it is important to combine the pre-equalizer with a digital IQ sample estimator, which derives the IQ values from the envelope signal without a down-modulation process. This paper provides a mathematical and architectural description and simulation results of the software-based IQ estimation and filter pre-equalizer.*

*Index Terms – Multi-antenna, MIMO, OFDM, direct conversion, filter pre-equalization, IQ estimation*

## I. Introduction

Multi-antenna systems enable high data-rates and a good system-performance [1], [2] if they can provide good signal accuracy already at the transmitter output [3]. This is problematical if a low-cost direct conversion architecture has been chosen, which offers a cheap and low-power implementation with the drawback of imperfect I- and Q-signal accuracies. Reasons for the imperfections can be imperfect analog base band filters and unbalanced I- and Q-branch amplification.
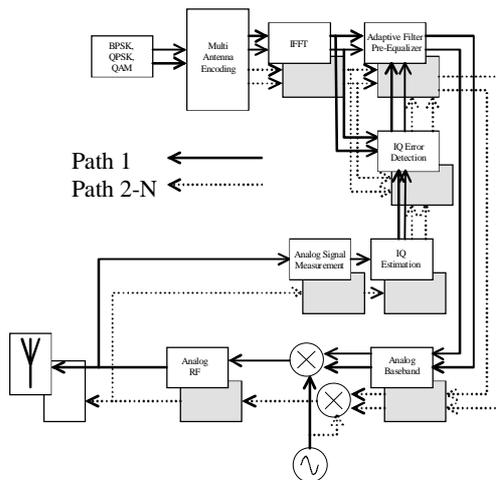


Figure I-1 Multi-antenna OFDM transmitter.

Hence analog signal imperfections have to be removed digitally, shown in Figure I-1. The transmitter generates first low rate QPSK or QAM constellation points, which will be mapped onto the different carriers [4]. The multi-antenna coding inserts e.g. diversity or spatial-multiplexing approaches [1], [2]. This is followed by the signal conversion from the frequency domain to the time domain. After the IFFT operation has been finished the pre-equalizer takes care about the analog filter imperfections [6], [7], [8]. This block is able to pre-modify the ideal digital data stream. The digital pre-modification effects will be compensated later by the introduction of the unwanted analog imperfections, such, that the final result at the antenna input port leads to an ideal signal. The digital pre-compensation block receives its correction coefficients from the error detection algorithm, which calculates the error based on the comparison between ideal and real I- and Q-samples. Hence an I- and Q-sample estimation from the RF-envelope signal is required. Typically this information is not available at this stage, because the RF signal does not provide an access to the base-band equivalent signals I and Q without a complete down-modulation process.

The purpose of this paper is to provide a simple IQ sample estimation combined with a digital filter pre-equalization algorithm. These algorithms will be

implemented in each branch of the multi-antenna transmitter. The digital components save costs, because cheap, non precise analog filters can be incorporated. Because of the multi-antenna approach it saves even more costs by implementing the algorithms via software to a Digital-Signal-Processor (DSP). The analog components will not change quickly their amount of imperfections during a certain period of time and hence an algorithm reuse of all compensation blocks via a DSP can have significant influence to the overall system complexity in terms of hardware gate count and power consumption.

## II. Digital IQ Estimation

In direct conversion architectures the I- and Q-branches are fed from the digital base band via two independent DACs to the analog base band. After separate low-pass filtering and appropriate amplification of each branch the up-conversion to the RF range takes place. In case of a multi-antenna system with $N$ – transmitter antennas this architecture has to be installed $N$ times. At the $N$ antenna outputs it is desired to have the best possible signal accuracy available. This can be reached by installing precise, but most probably expensive analog components for each I- and Q-branch and for each transmitter path. An advantageous alternative is the installation of low cost analog components with less precision and additionally digital compensation techniques to remove the analog imperfections via a cheap solution. Therefore I- and Q-signal extraction from the RF-envelope needs to be done. The extraction is required to estimate reliably the wanted IQ samples from the analog RF-envelope without a down-modulation process on the transmitter side.

In this paper the estimated IQ samples are used for the digital pre-equalization process. To pre-equalize the analog base-band filters there has to be employed filter imperfection estimation. Such an error detection could be done by subtracting the non-ideal IQ samples $\tilde{I}[n]$ and $\tilde{Q}[n]$ from the ideal IQ samples $I[n]$ and $Q[n]$.

$$\begin{aligned} e_I[n] = I[n] - \tilde{I}[n] \\ e_Q[n] = Q[n] - \tilde{Q}[n] \end{aligned} \tag{1}$$

In case of ideal output samples at the antenna port the differences between the wanted and the transmitted signals equals zero and no pre-equalization need to be activated. Assuming that there are imperfections present then the difference is unequal to zero in both branches from equation (1). To enable the required measurement the non-ideal IQ samples have to be extracted from the analog envelope signal. This will be done digitally by comparing two consecutive analog

and digital IQ pairs. The analog samples are measured at the antenna input port, the corresponding digital samples before the DAC operation. Two consecutive analog samples are described by equation (2). The amplitude $\left| A_a[n] \right|$ and the amplitude $\left| A_a[n-1] \right|$ are measured at the time instances $n$ and $n-1$. Only the left sides of both branches in equation (2) can be measured physically at the antenna input port. The elements of the right side have only theoretical meaning.

$$\begin{aligned} \left| A_a[n] \right| = \sqrt{\tilde{I}[n]^2 + \tilde{Q}[n]^2} \\ \left| A_a[n-1] \right| = \sqrt{\tilde{I}[n-1]^2 + \tilde{Q}[n-1]^2} \end{aligned} \tag{2}$$

At the same time it is necessary to measure the corresponding digital sample amplitudes. This is shown in equation (3).

$$\begin{aligned} \left| A_d[n] \right| = \sqrt{I[n]^2 + Q[n]^2} \\ \left| A_d[n-1] \right| = \sqrt{I[n-1]^2 + Q[n-1]^2} \end{aligned} \tag{3}$$

The digital amplitudes at the time instances $n$ and $n-1$ on the left side need to be calculated, because the I- and Q-branches provide $I[n], Q[n]$ and $I[n-1], Q[n-1]$ separately. For convenience further digital or analog component latency has been neglected.

The corresponding analog and digital amplitudes need to be compared. In ideal case the corresponding amplitudes should equal by omitting a certain constant amplification factor.

$$\begin{aligned} \left| A_d[n] \right| \overset{!}{=} \left| A_a[n] \right| \\ \left| A_d[n-1] \right| \overset{!}{=} \left| A_a[n-1] \right| \end{aligned} \tag{4}$$

Then the digital pre-equalization error detector from equation (1) would have indicated no error. But in practice the analog base band filters might add signal imperfections. To calculate the missing signals the following relationship will be taken into account.

$$\begin{aligned} \frac{\tilde{I}^2[n]}{\tilde{I}^2[n-1]} \overset{!}{=} \frac{I^2[n]}{I^2[n-1]} \\ \frac{\tilde{Q}^2[n]}{\tilde{Q}^2[n-1]} \overset{!}{=} \frac{Q^2[n]}{Q^2[n-1]} \end{aligned} \tag{5}$$

Equation (2), equation (3) and equation (5) can be used to re-formulate equation (4). This leads to

$$\tilde{I}[n] = s_I \cdot \sqrt{A_a[n]^2 - \tilde{Q}[n]^2} \tag{6}$$

and

$$\tilde{Q}[n] = s_Q \sqrt{\frac{Q[n]^2}{Q[n-1]^2} \cdot \frac{A_a^2[n-1] \cdot \frac{I[n]^2}{I[n-1]^2} - A_a^2[n]}{\frac{I[n]^2}{I[n-1]^2} - \frac{Q[n]^2}{Q[n-1]^2}}} \qquad (7)$$

First equation (7) needs to be solved and after that equation (6) can be taken into account. The signals $s_I$ and $s_Q$ provide the digital sample signs. They have been stored in parallel and it is assumed that the analog imperfections will not disturb the sign of the analog samples. This will be almost true, because one takes care about imperfect analog filters and no random channels.

But generally it is no problem if every now and then some wrong IQ estimates will occur. Because the estimation procedure has been designed to operate in conjunction with an adjustment feedback loop. The feedback loop employs a low-pass filtering process, which automatically removes the influence of wrong IQ decisions from the IQ estimate algorithm in equation (6) and (7). Such wrong estimates might happen because of the missing down-modulation on the transmitter side there is no information about the signal phase available. This missing phase information is not critical as long as the overall system imperfections are generated simply by imperfect analog components and not by a random channel. Figure II-1 provides the differences between the ideal and estimated I-values during a pre-equalizer's adaptation process. After all filter imperfections have been compensated by the pre-equalizer there has been left no differences between the ideal and estimated I-values.
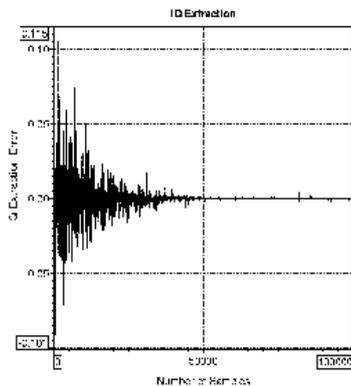
This section has presented a mathematical description of an IQ sample extraction algorithm from the analog RF-envelope. Without the need for a down-modulation process on the transmitter side it is possible to estimate the imperfect I- and Q-samples, which are required to enable the compensation techniques for direct-conversion front-end architectures.

## III. Filter Pre-Equalization

In this section there will be introduced an LMS based pre-equalizer [7], [8], which does not operate with complex coefficients, but with real ones. This is unusual but makes it possible to handle I-branch and Q-branch imperfections independently. The I-branch and Q-branch filter imperfections are generated by the analog base band filters, which are two real filters. The IQ amplitude error detection will be done via equation (1).

To update the pre-equalizer's filter coefficients successfully the gradient has to be calculated based on the approximated system identification [7]. The approximation of the analog filters will be simple tap-delay lines providing the same latency as the analog filters contain. Equation (8) provides the gradient of the LMS approach.

$$\hat{\underline{\nabla}} \left\{ \hat{E} \left\langle e^2[n] \right\rangle \right\} = -2 \cdot e[n] \cdot \underline{D}[n] \cdot \underline{h}^{\#}[n] \qquad (8)$$

The approximation-based gradient is updated on a sample-by-sample base and depends on the measured error value e[n] and the delayed input signal. The mentioned delay corresponds to the approximated analog filter peak. Figure III-1 provides the difference between the proposed pre-equalizer with an approximation-based gradient, a gradient based on an ideal system identification and a deterministic gradient.



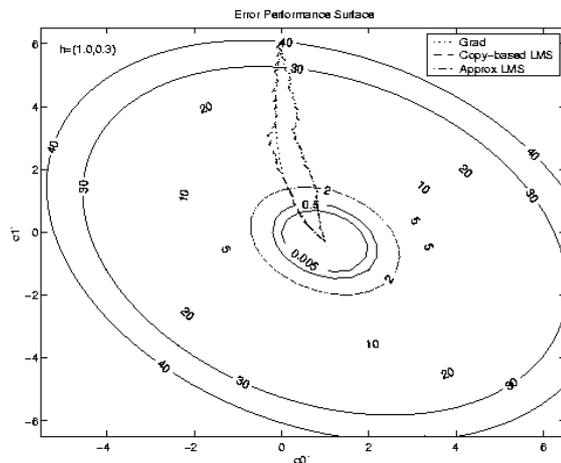Figure II-1 Estimated I-branch error during a pre-equalizer's adaptation process.



Figure III-1 Gradients on the level curve diagram.

The approximation-based gradient takes a different route but reaches the optimal filter vector as the other algorithms. Figure III-2 shows the three gradients from another camera position.
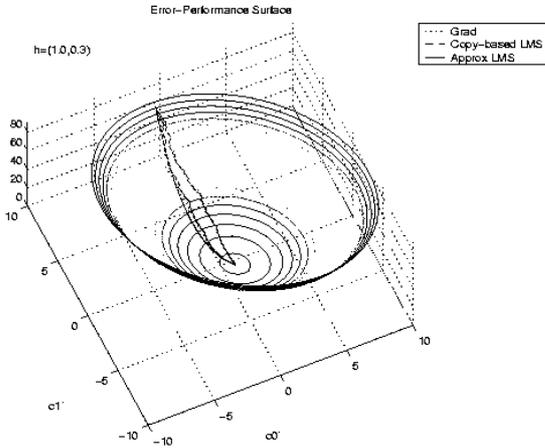


Figure III-2 Gradient behaviour from bird's eye view

Based on the gradient there can be calculated the pre-equalizer's coefficient update. There have to be calculated for both branches independent correction coefficients. This is described by equation (9). The new coefficients at the time n+1 will be calculated from the current coefficients at the time n and an additional addend.

$$\underline{c}_{I,Q}[n+1] = \underline{c}_{I,Q}[n] + \mu\, e_{I,Q}[n]\underline{D}_{I,Q}[n]\underline{h}_{I,Q}^{\#}[n] \qquad (9)$$

The addend consists out of four factors. First the constant $\mu$ describes the step width. The step width defines the loop accuracy, loop adaptation speed or loop bandwidth, respectively. Because the expected filter imperfections will not change over a very long period of time the loop bandwidth needs not to be large and hence the loop accuracy can be high. The second factor is the calculated error from equation (1). After that the product of the ideal input data matrix $\underline{D}$ and an approximation $\underline{h}^{\#}$ of the analog filters $\underline{h}_{I,Q}$ follows. The new coefficient vector leads to a better signal equalization and if the optimium adaptive filter vector has been reached the adaptation loop is in equilibrium.

Combined with the IQ estimation there can be build an adaptive filter pre-equalization system to enable low cost analog front-ends.

## IV. Software Architecture for further cost reduction

From the architecture point of view it will be advantageous to implement the algorithms as software code via a Digital-Signal-Processor [9]. The mathematical operations from equation (1), (6), (7) and (9) are good candidates to be handled by the DSP. This is true because the analog filter imperfections do not change quickly. Hence the IQ sample estimation, error calculation and the coefficient update need not to be done as quickly as practical possible. Changing the block based hardware implementation from Figure I-1 it is possible to end up with a much more flexible and cost-reducing architecture by employing a DSP.
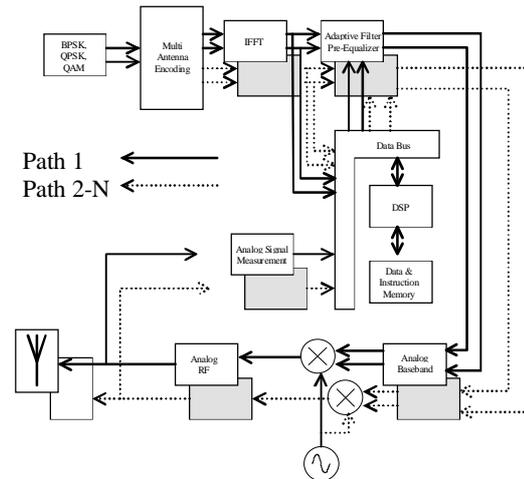


Figure IV-1 Software-based IQ estimation, pre-equalizer's error and coefficient update calculation.

Figure IV-1 shows a software-based transmitter part for the IQ sample estimation, the pre-equalizer's error detection as well as the coefficient update. The data bus establishes the connections between the DSP and the envelope input signal and the pre-equalizer's adaptive filters, respectively. The instructions for the different algorithms, which have been implemented via dedicated hardware in Figure I-1, are stored now in the instruction memory in Figure IV-1. Additional control SW, which is responsible to guarantee the correct order of the different algorithm operations, needs to be provided as well. Besides the instructions the DSP requires the data from the digital base band and the analog front-end. The information is stored in the data memory and used by the DSP instructions to calculate the new coefficient update.

Once the coefficients have been updated they can be provided via the bus to the pre-equalizer's adaptive filters. The filters are still implemented via dedicated HW because the signal pre-modification needs to operate on the base of the user data rate. From the instructions point of view the DSP could handle the adaptive filtering process as well. But in that case a significant higher processor clock rate needs to be

considered. Such a high clock rate might increase the power consumption of the DSP to an unwanted value.

## V. System Performance

This chapter shows a new analysis about the performance decrease by introducing sub-optimal analog filters and the corresponding signal improvements through the digital pre-equalization setup. There has been investigated an IEEE802.16a based OFDM system including 16-QAM and 64-QAM.
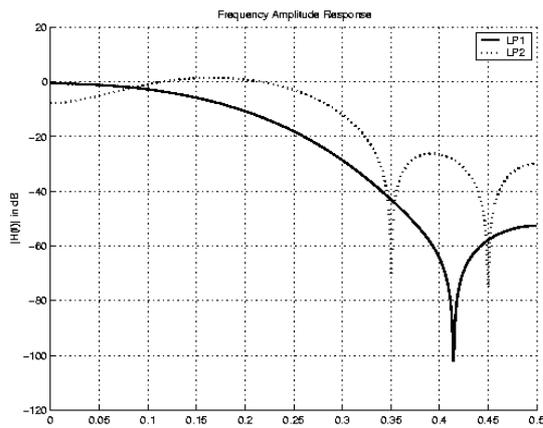
Figure V-1 Sub-optimal filter transfer functions.

Because of the cost reduction for the analog front-end one assumes imperfect filters. Based on the transfer functions from Figure V-1 there can be expected a significant decrease of the transmitted signal accuracy. Figure V-2 shows possible inaccuracies for a 16-QAM signal.
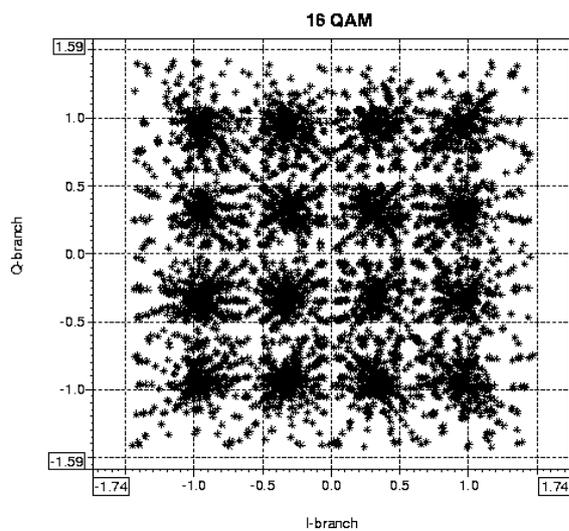
Figure V-2 Imperfect 16-QAM constellation diagram.

After the pre-equalization process has been enabled the imperfections are reduced significantly already by a 3-coefficient adaptive filter. Figure V-3 shows that the constellation points are much more precise but not perfect.
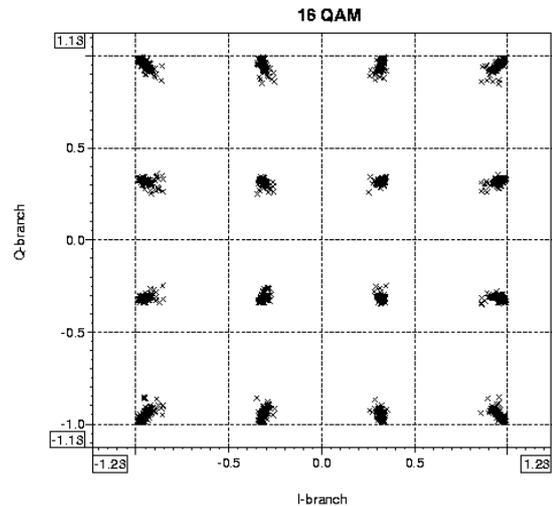
Figure V-3 3-coefficients pre-equalizer

By employing 19 coefficients perfect signal accuracy at the transmitter's output can be reached. This is shown in Figure V-4. Hence a digital adaptive filter can allow the use of low-cost, imperfect analog filters.
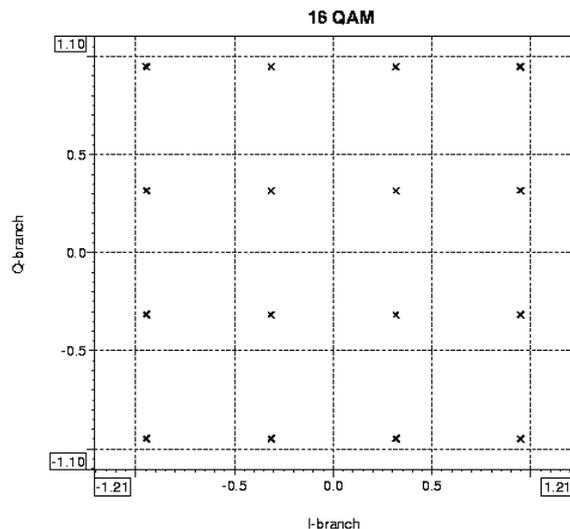
Figure V-4 Perfect pre-equalized 16-QAM signal.

Besides the signal accuracy it is possible to measure the imperfections via BER curves as well. Figure V-5 and Figure V-6 provide simulation results for 16-QAM and 64-QAM, respectively.
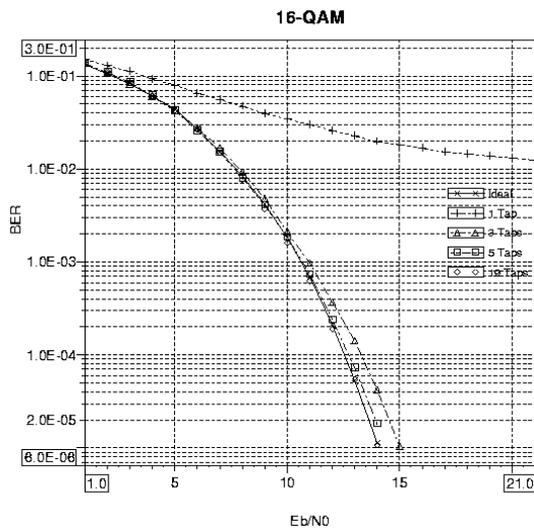
**16-QAM**



Figure V-5 16-QAM BER curves with different pre-equalizer coefficient numbers.

Non-frequency selective corrections employ only 1 coefficient and cannot remove the imperfect analog filter influences. They adjust just the signal's amplitude. A BER floor makes the overall transmitter performance pure. Increasing the number of pre-equalizer coefficients leads to better performances. In case of a 64-QAM there is a very high BER floor without corrections and also a 3-coefficients pre-equalizer suffers still significant losses. With 19 coefficients the desired performance is provided.
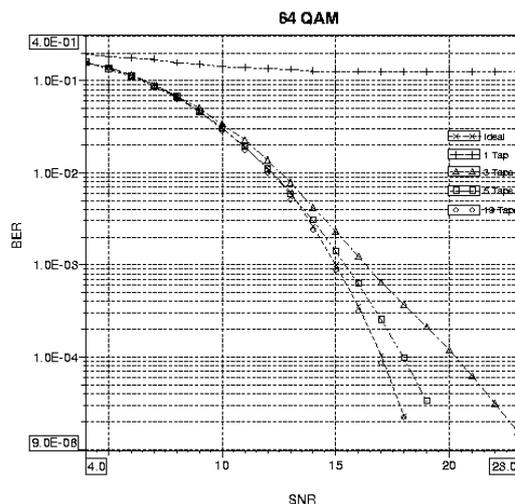
**64 QAM**



Figure V-6 64-QAM BER curves.

This chapter has shown that the overall system performance decreases significantly by introducing low-cost analog filters. Depending on the Euclidean distance a 64-QAM signal is much more sensitive against filter inaccuracies than 16-QAM signal. Finally a 19-coefficients pre-equalizer can remove the imperfections and high signal accuracy at the transmitter's output can be reached.

## VI. Conclusion

This paper introduces a cost-reduction model for multi-antenna transmitters. Because it is important to provide low-cost analog front-ends for multi-antenna systems there is a need to remove the imperfections of the low-cost analog base band filters. This can be done by an IQ sample estimation algorithm, which calculates the IQ symbols at the transmitter's antenna output without an extra down-modulation process. Feeding the estimates to the digital pre-equalizer the imperfections can be removed completely. By employing a software-based IQ estimation and pre-equalization setup a low gate count implementation for multi-antenna systems can be reached. IQ estimation and pre-equalizer achieves significant improvements for the BER. This leads to high system reliability although low-cost analog filters have been used.

### Acknowledgment

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