ITERATIVE SIGNAL PROCESSING FOR MITIGATION OF WIDEBAND ADC NONIDEALITIES IN COGNITIVE RADIO RECEIVER

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ABSTRACT

Large input signal dynamics is an essential problem in modern wideband communication receivers such as cognitive radios. If the receiver front-end and analog-to-digital interface cannot respond to varying conditions, a high amount of nonlinear distortion is caused due to the clipping. This paper proposes a robust digital signal processing method to iteratively remove unintentional clipping distortion in OFDM receivers. Using computer simulations it is shown that the symbol error ratio of a heavily clipped signal in a fading channel situation can be reduced practically to the level of an equivalent non-clipped signal. In other words, the proposed method can remove all the essential clipping distortion.

1. INTRODUCTION

In modern wideband cognitive radio receivers, unintentional signal clipping is a potential problem due to the highly variable signal dynamics. This is partially stemming from the high peak-to-average power ratio (PAPR) of modern communication waveforms. Other crucial cause for the variable signal dynamics in a cognitive radio front-end is high-power transmissions of adjacent-channel users. This paper proposes a novel decision-aided iterative approach for digitally compensating unintentional clipping of analog-to-digital (A/D) converters in a scenario where multiple users are transmitting orthogonal frequency division multiplexing (OFDM) signals. The focus is particularly on the case where the reception of a weak signal is compromised due to the high amount of clipping distortion stemming from a considerably stronger adjacent-channel signal.

In the current literature, there are just a few digital signal processing methods for reducing unintentional signal clipping occurring *on the receiver side* [1], [2], [3]. All of them are rather general methods and do not take into account the specific properties of OFDM signals. On the other hand, OFDM-related clipping mitigation methods in the current literature focus on deliberate clipping to reduce signal PAPR *on the transmitter side* [4], [5], [6]. These methods assume that the exact clipping level is known, which is a fair assumption when the clipping is intentional. However, this paper proposes a method against unintentional clipping, which means that also the occurred clipping level has to be estimated. Another challenge in this scenario is the channel estimation since the clipping occurs after the signal has propagated through the radio channel.

The proposed method is based on the idea that bit decisions can be exploited to create an estimate of the clipping distortion in the received signal. In the literature, decision-based approaches have been found to be successful for mitigating many RF impairments, e.g., phase noise [7], [8]. The method proposed in this paper is using the PAPR reduction scheme discussed in [5], [6] as a starting point, and extends that to the challenging case of receiver clipping in the presence of neighboring channels. This paper shows through computer simulations that the proposed method is robust against different kind of estimation errors and is able to extensively remove clipping distortion.

The rest of the paper is organized as follows. Section 2 gives the basics of modeling the clipping of OFDM signals relying on the results provided in the current literature. Based on this signal modeling, Section 3 proposes a novel clipping compensation approach. Its performance is analyzed with computer simulations in Section 4 and finally Section 5 draws conclusions.

2. CLIPPING OF OFDM SIGNALS

One OFDM symbol with $N = JN_A$ subcarriers (SC) has N_A active SCs and J denotes the oversampling factor. All SC indices are defined as a set $\Omega = \{-N/2, -N/2+1, ..., N/2-1\}$. The active SCs contain a sequence of complex data symbols X_k, $k \in \Omega_A = \{-N_A/2, ..., -1, 1, ..., N_A/2\}$. Zero padding is used so that $X_k = 0,\ k \in \Omega \setminus \Omega_A$. Now the baseband OFDM symbol is defined as

$$x(t) = \frac{1}{\sqrt{N}} \sum_{k=-N/2}^{N/2-1} X_k e^{j2\pi kt/T_s}, \quad 0 \le t \le T_s, \tag{1}$$

where T_s is the OFDM symbol duration. This model does not include the cyclic prefix (CP), which is required for a successfully working OFDM system in frequency-selective fading scenarios. The CP is omitted from the mathematical notation of the paper to make it simpler. This is justified since the CP does not have any significance in the presented analysis. However, due to the importance of CP in practical systems, it is implemented in the simulations shown in Section 4.

This paper assumes that the signal clipping in the A/D conversion stage can be approximated with a zero-symmetric hard clipping. Denoting a general received signal by $y(t) = y_I(t) + jy_Q(t) = h(t) * x(t) + w(t)$, where $h(t)$ is the channel impulse response and $w(t)$ denotes additive noise, the clipped received signal $\tilde{y}(t) = \tilde{y}_I(t) + j\tilde{y}_Q(t)$ is given by

$$\tilde{y}_I(t) = \begin{cases} y_I(t), & |y_I(t)| < V_0 \\ V_0, & y_I(t) \ge V_0 \\ -V_0, & y_I(t) \le -V_0 \end{cases} \tag{2}$$

for the I branch and similarly for the Q branch. Here V_0 is the maximum acceptable input level of the A/D converter and it is assumed to be equal for both branches. In practice, it is convenient

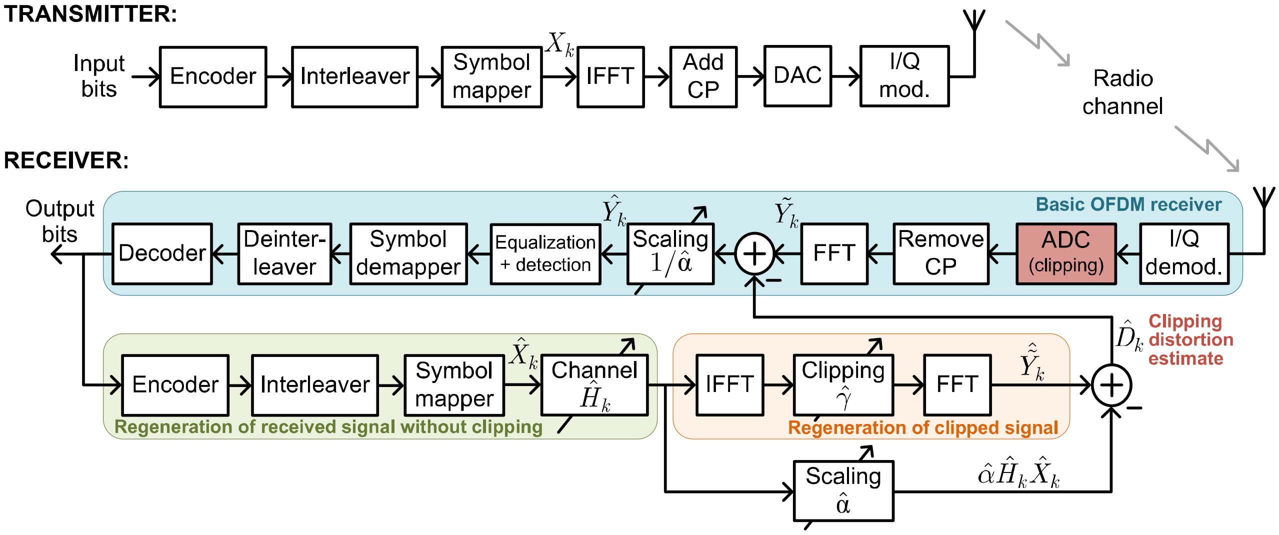

Figure 1 – Block diagram of the overall OFDM system for one transmitter and receiver. ADC on the receiver side causes undesired clipping distortion, which can be removed using the proposed method. Capital letters denote frequency domain signals at different stages.

to define the clipping level as a ratio of V_0 and the average power level of the complex signal before clipping P_{in}. Therefore, the clipping level γ is defined as

$$\gamma = \frac{V_0}{\sqrt{P_{\text{in}}/2}}. \tag{3}$$

However, it is not very simple to directly apply the clipping model in (2) for clipping compensation purposes. Therefore, this paper models clipping distortion using Bussgang's theorem [9]. This has already been shown to be an appropriate approach [10], [11]. Clipping is a memoryless nonlinearity and assuming $y(t)$ to be Gaussian, it is possible to write according to Bussgang's theorem that the clipped version is then

$$\tilde{y}(t) = \alpha y(t) + d(t). \tag{4}$$

This means that the clipping attenuates the original signal with a factor α and additionally causes additive clipping distortion $d(t)$, which is uncorrelated with $y(t)$. It is shown in [12] that the attenuation factor is

$$\alpha = \text{erf}\left(\frac{\gamma}{\sqrt{2}}\right) \tag{5}$$

in the case where $y_I(t)$ and $y_Q(t)$ are Gaussian.

3. DECISION-AIDED ITERATIVE CLIPPING COMPENSATION APPROACH

Subsection 3.1 introduces the clipping compensation algorithm in general. After that Subsection 3.2 discusses about details of implementation in multi-user scenarios.

3.1 Algorithm Description

The whole OFDM transmission chain is illustrated with a block diagram in Figure 1. For simplification, the figure shows only one transmitter although the proposed compensation algorithm on the receiver side is able to jointly process the data of several users as is discussed in Subsection 3.2. In order to simplify the notation of the algorithm description, only one OFDM symbol duration is considered in the equations. This is reasonable since the clipping compensation can be performed individually for every OFDM symbol.

If the received signal gets clipped in the A/D converter (ADC) due to the improper input signal conditioning, the resulting clipping distortion can be mitigated as shown in Figure 1. The basic idea is that the received signal is first detected in a traditional manner. Then, an estimate of the received signal with and without clipping is generated based on the obtained bit decisions. An estimate of the clipping distortion can be found by subtracting the estimate of the non-clipped signal from the clipped one. Now, this estimated clipping distortion can be removed from the original received signal and the signal is detected again. In the following paragraphs, the compensation approach is described in more detailed manner and Table 1 summarizes the key steps.

In ideal case, the received signal would be digitized without any clipping. After A/D conversion, CP removing and taking FFT, this frequency-domain signal can be expressed as

$$Y_k = H_k X_k + W_k, \quad k \in \Omega, \tag{6}$$

where H_k and W_k are the channel gain and the additive white Gaussian noise term for the kth SC, respectively. On the other hand, in case of clipping, the received signal in frequency domain (see Figure 1) is

$$\begin{aligned} \tilde{Y}_k &= \alpha Y_k + D_k \\ &= \alpha(H_k X_k + W_k) + D_k, \quad k \in \Omega \end{aligned} \tag{7}$$

as defined according to Bussgang's theorem in (4). Before starting the clipping compensation procedure, the data symbols are detected using $\tilde{Y}_k$, $k \in \Omega_A$. This is illustrated with the "equalization + detection" block in Figure 1. This stage also includes channel estimation and compensation using pilot SCs since it is an essential requirement for the symbol detection. The channel gain estimates for active SCs are denoted with $\hat{H}_k$, $k \in \Omega_A$. Finally, the output bits are obtained after deinterleaving and decoding.

In the first stage of the clipping compensation, the detected output bits are used to regenerate the transmitted OFDM signal. Since this is an estimate of the sent data, the frequency-domain signal is denoted as $\hat{X}_k$, $k \in \Omega$ in Figure 1. The signal is then exposed to the estimated channel and clipping process in order to regenerate the clipped received signal. The clipping regeneration is performed in time domain using the estimated clipping level $\hat{\gamma}$, which can be obtained by estimating (3), i.e., by checking the absolute maximum level of the signal after ADC $(\hat{V}_0)$ and calculating the average power level of $\hat{X}_k,\ k \in \Omega\ \ (\hat{P}_{\text{in}})$. The time domain clipping process essentially creates a signal that can be expressed in frequency domain as $\hat{\tilde{Y}}_k = \hat{\alpha}\hat{H}_k\hat{X}_k + \hat{D}_k,\ k \in \Omega_A$, where $\hat{\alpha}$ is the estimate of the attenuation factor. From $\hat{\tilde{Y}}_k,\ k \in \Omega_A$ it is possible to extract the clipping distortion estimate $\hat{D}_k,\ k \in \Omega_A$ by removing the wanted signal part. The lowest branch in Figure 1 is illustrating this by generating a non-clipped but attenuated estimate of the received signal, namely $\hat{\alpha}\hat{H}_k\hat{X}_k,\ k \in \Omega_A$, where $\hat{\alpha}$ is calculated from (5) using $\hat{\gamma}$. Now the estimate of the clipping distortion $\hat{D}_k,\ k \in \Omega_A$ can be obtained as

$$\hat{D}_k = \hat{\tilde{Y}}_k - \hat{\alpha}\hat{H}_k\hat{X}_k, \quad k \in \Omega_A. \tag{8}$$

The out-of-band clipping distortion is ignored since the processing focuses only to active SCs Ω_A as implied in (8). Finally, the estimated clipping distortion is removed from the received signal $\tilde{Y}_k,\ k \in \Omega_A$ to provide cleaner signal

$$\hat{Y}_k = \frac{1}{\hat{\alpha}}\left(\tilde{Y}_k - \hat{D}_k\right), \quad k \in \Omega_A \tag{9}$$

for a new round of channel estimation and symbol detection. If (7) is substituted in (9), the estimate of the received signal without clipping can be written as

$$\hat{Y}_k = \frac{\alpha}{\hat{\alpha}}(H_k X_k + W_k) + \frac{1}{\hat{\alpha}}\left(D_k - \hat{D}_k\right), \quad k \in \Omega_A. \tag{10}$$

From this form it is easy to see that having exactly correct estimates $\hat{\alpha}$ and $\hat{D}_k,\ k \in \Omega_A$ will produce totally non-clipped signal, which is equivalent to (6).

Errors in the first symbol detection cause that all the clipping distortion is not removed on the first round. By iterating, the compensation process will produce better and better estimate of the clipping distortion and therefore symbol detection results improve. In every iteration, new channel estimates $\hat{H}_k,\ k \in \Omega_A$, clipping level $\hat{\gamma}$, and attenuation factor $\hat{\alpha}$ are calculated.

3.2 Multi-User Aspects

The proposed compensation approach is applicable both in uplink (UL) and downlink (DL) scenarios, but there are certain differences in case of multiple users. In DL direction, the sequence $X_k,\ k \in \Omega_A$ consists of symbols belonging to several users. In order to compensate clipping distortion in the best possible manner, the mobile receiver should be able to process the symbols of all users and estimate the channel over the whole band. This increases the amount of required processing compared to the case where the mobile concentrates only on its dedicated band.

Also in UL direction the sequence $X_k,\ k \in \Omega_A$ contains symbols of several users. Additionally, channel estimates $\hat{H}_k,\ k \in \Omega_A$ have to be obtained piecewise since different users have different channels. The UL scenario is more feasible than DL from the implementation point of view, because the base station processes the signals of every user in any case even if there is no clipping compensation. In addition, base stations have more computing power making more complex DSP algorithms viable.

4. SIMULATION RESULTS

Presented computer simulations consider an UL scenario of two individual users whose signals propagate through different realizations of extended ITU-R Vehicular A channel [13]. It is assumed that the channels stay static over ten OFDM symbols. The average received signal power for User 1 is 15 dB less than for User 2. Both signals contain 512 active SCs of which every eighth SC is used as a pilot. Thus for the composite signal processed on the receiver side $N_A = 1024$. It is also assumed that the oversampling factor $J = 2$. The sampling rate in the simulations is 30.72 MHz, SC spacing is 15 kHz, and the SC modulation for both users is 16-QAM. Channel coding and interleaving are not used in the simulations since they are not on the main focus of this paper. Thus only raw symbol decisions are used in the iterative processing. Figure 2 illustrates the overall simulation scenario in time and frequency domains when $\gamma = 6$ dB and PAPR for the overall waveform is 10.5 dB.

Table 1 – The proposed receiver clipping compensation algorithm.

Initialization: for $k \in \Omega_A$

1. Set $i = 0$, $\hat{\alpha}^{(0)} = 1$ and $\hat{D}_k^{(0)} = 0$
2. Obtain $\hat{H}_k^{(i)}$ from $\hat{Y}_k^{(i)} = \frac{1}{\hat{\alpha}^{(i)}}\left(\tilde{Y}_k - \hat{D}_k^{(i)}\right)$
3. Detect/decode $\hat{Y}_k^{(i)}$ after channel compensation

Iteration: Increase i by one and do the following steps for $k \in \Omega_A$

1. Create $\hat{X}_k^{(i)}$ based on the decoded bits
2. Estimate $\hat{\gamma}^{(i)}$ and $\hat{\alpha}^{(i)}$
3. Clip $\hat{H}_k^{(i-1)}\hat{X}_k^{(i)}$ in *time domain* to create $\hat{\tilde{Y}}_k^{(i)}$
4. Calculate $\hat{D}_k^{(i)} = \hat{\tilde{Y}}_k^{(i)} - \hat{\alpha}^{(i)}\hat{H}_k^{(i-1)}\hat{X}_k^{(i)}$
5. Remove distortion, i.e., $\hat{Y}_k^{(i)} = \frac{1}{\hat{\alpha}^{(i)}}\left(\tilde{Y}_k - \hat{D}_k^{(i)}\right)$
6. Obtain $\hat{H}_k^{(i)}$ from $\hat{Y}_k^{(i)}$
7. Detect/decode $\hat{Y}_k^{(i)}$ after channel compensation

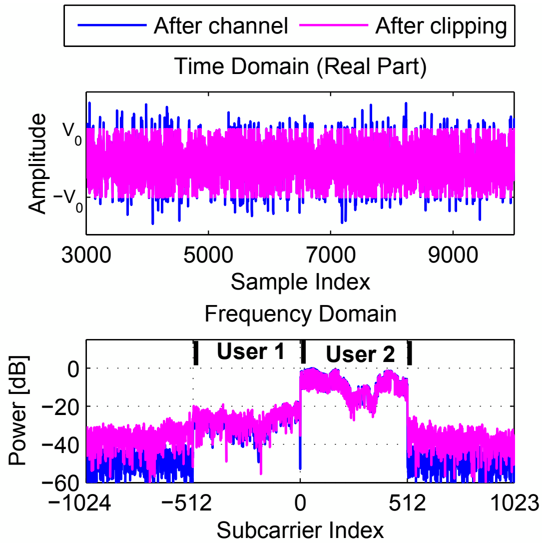

Figure 2 – Upper part: piece of the real branch signal illustrated in time domain before and after clipping, the clipping level being 6 dB above the average power level of the signal. Lower part: spectrum illustration of the signal before and after clipping. Frequency bands corresponding to Users 1 and 2 are denoted in the spectrum.

Raw symbol error ratios (SER) for User 1 and 2 as a function of compensation iteration rounds are presented in Figure 3 for the scenario described earlier. All the results are averaged over 10,000 OFDM symbols, i.e., 1,000 independent channel realizations. Iteration round 0 refers to the received uncompensated signal. Using ideal channel estimates and non-clipped signal, the lower bound for SER in this scenario can be found. As shown in Figure 3, the proposed compensation algorithm is able to remove almost all the clipping distortion from the signal in case of perfect channel knowledge. There is a difference between SERs of User 1 and 2 because the signal of User 1 is more attenuated and therefore more affected by the noise. Here the average received signal-to-noise ratio (SNR) is defined as

$$SNR_{k,u} = \frac{E\left[|H_k X_k|^2\right]}{E\left[|W_k|^2\right]}, \quad k \in \Omega_{A,u}, \tag{11}$$

where u is user index. For User 1 $\Omega_{A,1} = \{-N_A/2,\ldots,-1\}$ and for User 2 $\Omega_{A,2} = \{1,\ldots,N_A/2\}$. In the case shown in Figure 3, $SNR_{k,1} = 26\,\mathrm{dB}$, $k \in \Omega_{A,1}$ and $SNR_{k,2} = 41\,\mathrm{dB}$, $k \in \Omega_{A,2}$.

In the simulations, the channel estimates for the data SCs are calculated using linear interpolation between pilot SCs. The acquired channel estimates are heavily affected by the clipping distortion and hence it takes several iterations to minimize SER. However, the proposed algorithm is robust enough to successfully remove most of the clipping distortion although there are channel estimation errors. In Figure 3, the performance gap between the non-clipped results with and without ideal channel knowledge is due to the limited accuracy of the used channel estimation method (pilot interpolation), which is not in any way related to the clipping phenomenon.

As more and more clipping distortion can be removed from the received signal by every iteration, also the channel estimates become more accurate. This is illustrated in Figure 4 where the mean square error (MSE) of the channel estimates is plotted subcarrier-wise. In this particular simulation scenario, the proposed algorithm is able to decrease the MSE approximately by 15 dB. The MSE after the compensation is very close to the ideal performance of the used channel estimation approach, which is illustrated with the MSE plot of non-clipped signal in Figure 4. Noticeable edge behavior in the figure is stemming from the applied channel estimation scheme. The distinct MSE peaks are due to the lack of pilot SCs on the right side of both user bands. The channel estimate provided by the last pilot SC is used as such for the remaining data SCs.

Another way to present the performance of the proposed clipping compensation algorithm is subcarrier-wise clipping distortion ratio (CDR). It describes the ratio of received signal power and clipping distortion power. Hence, the definition is

$$CDR_k = \frac{E\left[|Y_k|^2\right]}{E\left[\left|\hat{Y}_k - Y_k\right|^2\right]}, \quad k \in \Omega_A. \tag{12}$$

In Figure 5, the CDR results are shown for the previously described case before and after 10 iterations of the clipping compensation.

Especially on the first iteration round, the clipping distortion heavily affects the channel estimation accuracy which leads to symbol detection errors. If there are a lot of detection errors, the clipping distortion estimate is poor and it takes a considerable amount of iterations to make the received signal any cleaner. This can be seen from Figure 6, where SER results are plotted as a function of clipping level after ten iterations of the clipping compensation. It can be concluded from Figure 6 that heavy clipping causes

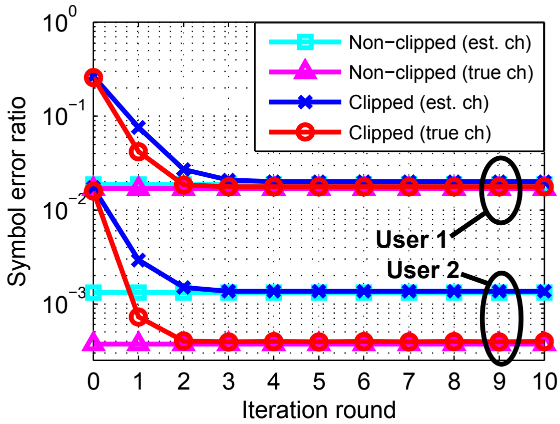

Figure 3 – SERs as a function of compensation iteration rounds for Users 1 and 2 when the clipping level is 6 dB over the average power level of the signal and average received SNR is 26 dB for User 1 and 41 dB for User 2. Iteration round 0 refers to uncompensated signal.

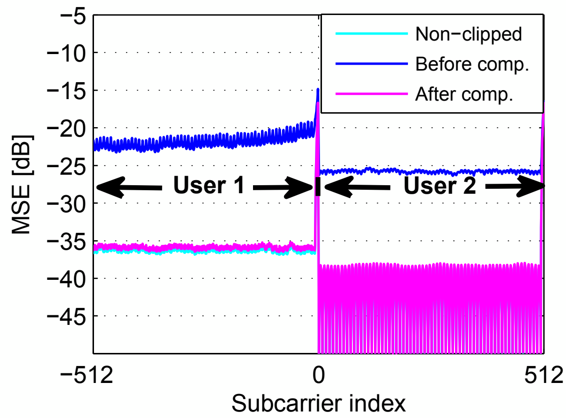

Figure 4 – Subcarrier-wise mean square error of channel estimates for both users before clipping compensation and after ten iterations of the proposed compensation. MSE for the non-clipped signal gives lower bound for the used channel estimation approach.

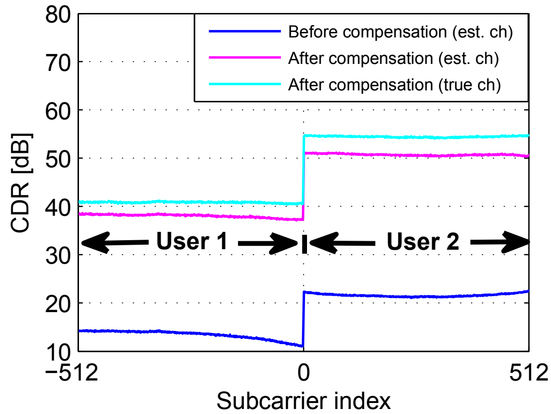

Figure 5 – Subcarrier-wise clipping distortion ratio for the case where clipping level is 6 dB over the average signal power level.

the channel estimation accuracy to decrease so much that the clipping compensation algorithm is unable to remove all the clipping distortion within 10 iterations. However, if the channel is known perfectly, the proposed clipping compensation algorithm performs better in extreme clipping situations. This is also verified by Figure 7, which illustrates SERs after 10 iterations of the clipping compensation as a function of average received SNR of User 1. On average, the received SNR of User 2 is 15 dB more. In addition to channel estimation accuracy, also the accuracy of the clipping level estimate $\hat{\gamma}$ has effect on the compensation performance. However, the effect is rather small in the simulations, if the error in $\hat{\gamma}$ is within $\pm 0.5\,\text{dB}$. This conclusion was obtained from a simulation, where $\gamma = 6\,\text{dB}$ and the value of $\hat{\gamma}$ was intentionally varied.

5. CONCLUSION

This paper proposed a novel iterative method for removing unintentional clipping occurring in the A/D converter of a wideband OFDM receiver. The method was shown to be able to remove almost all the clipping distortion even in extreme conditions and therefore recover the performance to the level of non-clipped signals. It was found out that under heavy clipping the compensation performance is mostly limited by the accuracy of the channel estimation. In the future, it should be studied, if it is possible to take the clipping phenomenon better into account in the channel estimation process.

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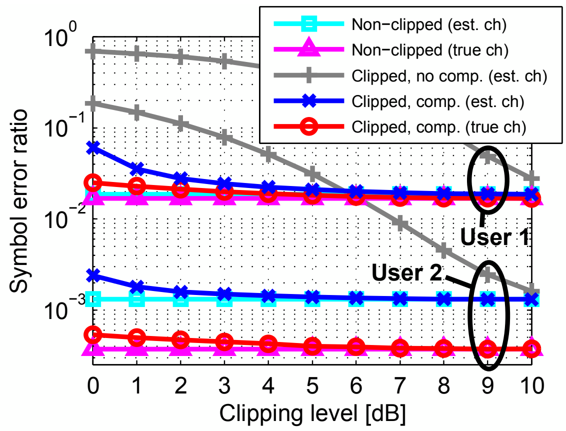

Figure 6 – SERs as a function of clipping level for Users 1 and 2 after 10 iterations of clipping compensation, average received SNR being 26 dB for User 1 and 41 dB for User 2.

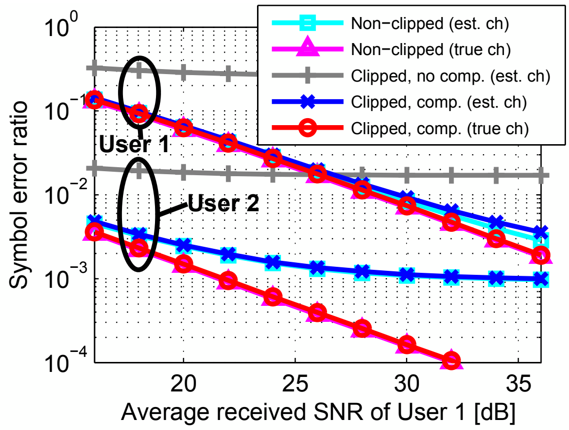

Figure 7 – SERs as a function of average received SNR of the weaker signal (User 1) after 10 iterations of clipping compensation, clipping level being 6 dB. The average received SNR for User 2 is 15 dB higher than for User 1.