MITIGATION OF CLIPPING IN SENSORS

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ABSTRACT

One major source of nonlinear distortion in analog-to-digital converters (ADCs) is clipping. The problem introduces spurious noise across the bandwidth of the sampled data. Prior works recover the signal from the acquired samples by relying on oversampling or on the assumption of vacant frequency bands and on the use of sparse signal representations. In this work, we propose a different approach, which uses two streams of data to mitigate the clipping distortion. Simulation results show an SNR improvement of 9dB, while the conventional approaches may even degrade the SNR in some situations.

Index Terms— ADC nonlinearity, clipping, spectrum sensing.

1. INTRODUCTION

In the design of Analog-to-Digital (ADC) converters, one important source of nonlinear distortion is clipping. The distortion occurs when the amplitude of the input signal exceeds or saturates the input range of the ADC. In this case, the sampled value is the max or min value of the ADC's input range (depending on whether the signal value is positive or negative). This impairment creates spurious noise across the entire bandwidth in the sampled data. The trend towards using wideband receivers in applications, such as spectrum sensing and geolocation [1–3], further accentuates the problem when multiple signals are digitally sampled together. Ideally, the gain of the receiver should be tuned such that the down-converted signals utilize the full dynamic range of the ADC. However, it is generally non-trivial to determine the maximum amplitude of the combined signals. Hence, it is difficult to optimize the dynamic range and prevent clipping distortions.

There are several prior works that proposed schemes to mitigate the clipping problem. Typically, these works use some of the following assumptions:

- The signal is oversampled.
- The signal is bandlimited or it contains some known vacant frequency bands.
- The signal has a sparse representation.
- The distorted signal can be modeled in terms of some polynomial representation.

For example, in [4], the proposed solution applies polynomial spline interpolation, followed by sinc interpolation, to the distorted samples. This approach requires oversampling and both two and four times oversampling rates were considered. In [5, 6], the authors exploit prior information about vacant frequency bands. Reference [5] assumes oversampling is used and exploits some out-of-band constraints. Reference [6] assumes that there are some empty subcarriers in OFDM signals at the outskirt of the OFDM spectrum. Both works [5, 6] employ an objective function that minimizes the noise power in the empty frequency bands. In [7–9], the authors examine the case when a bandlimited signal is oversampled and some samples are *lost*. If we treat the clipped samples as lost samples, the results in these works can also be applied to the current problem.

In [10], the proposed solution uses frame-based processing and sparse representation to model an audio signal and its clippings. Subsequently, a constrained matching pursuit algorithm is used to estimate the signal. In [11], the authors use a reweighted norm-1 and a trivial pursuit algorithm to remove clipping distortions when the signal consists of multiple sinusoidals. These algorithms are motivated by ideas from compressive sensing. In [12], the authors study the recovery of signals that have a sparse representation. One of the cases that they studied is clipping. They provide some recovery guarantees based on uncertainty relations between pairs of dictionaries.

In [13, 14], the authors use polynomial modeling to describe the clipping operation. The clipping distortions in the polynomial model are the higher order terms with some unknown coefficients. Adaptive filtering is used to estimate the unknown coefficients while minimizing the power of some desired target frequency band.

1.1. Motivation and contribution

The aforementioned approaches rely on the following ingredients: oversampling, knowledge of some out-of-band frequency bands, or some knowledge about the structure of the signal. The works also assume that they have access to only one stream of data.

Now consider the case in spectrum sensing or geolocation applications where data from multiple sensors are usually available. In this situation, it should be advantageous to exploit the multiple stream of data to mitigate clipping distortions. This paper addresses this problem and proposes a solution method using two streams of data, and that does not require the same assumptions as listed for the prior works.

2. PROBLEM FORMULATION

2.1. Data model

Suppose there are L sensors and K transmitters, which are distributed over some geographic location. Assume that the transmission medium is a flat fading channel. The signal transmitted by the k-th transmitter at time t is denoted by $s_k(t)$. This signal travels towards sensor ℓ and arrives there with delay $\tau_{k,\ell}$ units of time and with attenuation $A_{k,\ell}$, namely,

This work was supported in part by NSF grant CCF-1011918.

$$r_{k,\ell}(t) = A_{k,\ell} s_k(t - \tau_{k,\ell}) \quad (1)$$

First, assuming that there is no clipping, sensor ℓ will sample the sum of the signals from all transmitters at nT_s:

$$x_\ell[n] = \sum_{k=1}^{K} r_{k,\ell}(nT_s) + v_\ell(nT_s), \quad \ell = 1, \ldots, L \quad (2)$$

where $v_\ell(nT_s)$ is the noise at the ℓ-th sensor. In the clipping problem, we model the sampled data $\check{x}_\ell[n]$ as

$$\check{x}_\ell[n] = \begin{cases} -\text{CL} & \text{if } x_\ell[n] \leq -\text{CL} \\ x_\ell[n] & \text{if } -\text{CL} < x_\ell[n] < \text{CL} \\ \text{CL} & \text{if } x_\ell[n] \geq \text{CL} \end{cases} \quad (3)$$

where the clipping level CL is the maximum absolute input value of the ADC. We assume clipping occurs when $|\check{x}_\ell[n]| = \text{CL}$.

Figure 2 gives an example of the clipping distortion on a received signal in both the time and frequency domains. The original received signal consists of 2 bandlimited signals from 2 transmitters as shown in the bottom left plot. After clipping, the spectrum (in the bottom right plot) shows the distortions, which overlap with the original spectrum. The details for this example are in Section 2.3.

We assume that the frequency bands, Ω_k, used by the transmitters do not overlap. Using the discrete-time Fourier transform (DTFT) $\mathcal{F}$, the time-delayed and attenuated signal (1) can be represented as:

$$\mathcal{F}\left\{A_{k,\ell} s_k(nT_s - \tau_{k.\ell})\right\} = A_{k,\ell} e^{-j\omega\tau_{k,\ell}} S_k(e^{j\omega}) \quad (4)$$

where $S_k(e^{j\omega})$ is the DTFT of the original signal. Now, suppose there is some clipping at the ℓ-th sensor. We denote the indices of the clipped samples by $\{\phi_1^\ell, \phi_2^\ell, ...\phi_{M_\ell}^\ell\}$ and collect them into a vector Φ^ℓ whose cardinality is $\left|\Phi^\ell\right| = M_\ell$. Then, the desired unclipped samples $x_\ell[n]$ can be represented in vector form as

$$x_\ell = \check{x}_\ell + B_\ell e_\ell$$

$$x_\ell = \begin{bmatrix} x_\ell[1] \\ \vdots \\ x_\ell[N] \end{bmatrix}, \check{x}_\ell = \begin{bmatrix} \check{x}_\ell[1] \\ \vdots \\ \check{x}_\ell[N] \end{bmatrix}, e_\ell = \begin{bmatrix} e_\ell[1] \\ \vdots \\ e_\ell[M_\ell] \end{bmatrix} \quad (5)$$

and B_ℓ is a matrix where

$$B_\ell = \begin{bmatrix} b_{\ell,1} & \ldots & b_{\ell,M_\ell} \end{bmatrix}$$
$$b_{\ell,m}[n] = \begin{cases} -1, & \text{if } n = \phi_m^\ell \text{ and } \check{x}_\ell[n] = -\text{CL} \\ 1, & \text{if } n = \phi_m^\ell \text{ and } \check{x}_\ell[n] = \text{CL} \\ 0, & \text{otherwise} \end{cases} \quad (6)$$

Moreover, e_ℓ is an unknown vector representing the absolute error between the desired and clipped sample; (i.e., $e_\ell[m] = |x_\ell[m] - \check{x}_\ell[m]|, m \in \Phi^\ell$). Now, recall that the k-th transmitted signal lies in some frequency band denoted by Ω_k. We can represent each band using the N-point DFT matrix. Since each row of the DFT matrix corresponds to some frequency component of the sampled data, we use W_k to denote the rows of the DFT matrix that correspond to Ω_k. We use (5) to write:

$$W_k x_\ell = W_k(\check{x}_\ell + B_\ell e_\ell), \quad k = 1, ..., K \quad (7)$$

2.2. Relation between sensors

In cooperative sensing, we assume that the data can be shared among the sensors. Note from (1) and (2) that the sensors receive time-delayed and attenuated versions of the same transmissions. From (4), the time-delayed and attenuated versions can be expressed as some phase and amplitude change in the frequency domain. Now, suppose two sensors are labelled as the ℓ-th and p-th sensors, and both receive the transmission from the k-th transmitter. The received signals by the ℓ-th and p-th sensor can be expressed as $A_{k,\ell} s_k(nT_s - \tau_{k,\ell})$ and $A_{k,p} s_k(nT_s - \tau_{k,p})$, respectively. Their DTFT representations are

$$\begin{aligned} \mathcal{F}\left\{A_{k,\ell} s_k(nT_s - \tau_{k,\ell})\right\} &= A_{k,\ell} e^{-j\omega\tau_{k,\ell}} S_k(e^{j\omega}) \\ \mathcal{F}\left\{A_{k,p} s_k(nT_s - \tau_{k,p})\right\} &= A_{k,p} e^{-j\omega\tau_{k,p}} S_k(e^{j\omega}) \end{aligned} \quad (8)$$

We denote the relative amplitude ratio and the time-difference-of-arrival (TDOA) of the received signal as $\alpha_{k,\ell,p} = A_{k,\ell}/A_{k,p}$ and $\tau_{k,\ell,p} = \tau_{k,p} - \tau_{k,\ell}$, respectively. Therefore, the ℓ-th and p-th sensors' data in (7) are related with the k-th transmitter's signal as

$$\begin{aligned} W_k x_\ell &\approx \alpha_{k,\ell,p}\, Q_{k,\ell,p} x_p \\ W_k(\check{x}_\ell + B_\ell e_\ell) &\approx \alpha_{k,\ell,p}\, Q_{k,\ell,p}(\check{x}_p + B_p e_p) \end{aligned} \quad (9)$$

where

$$Q_{k,\ell,p} \triangleq \text{diag}\left\{\begin{bmatrix} e^{j\omega_{k,1}\tau_{k,\ell,p}} \\ \vdots \\ e^{j\omega_{k,R_k}\tau_{k,\ell,p}} \end{bmatrix}\right\} W_k, \quad (10)$$

and $\{\omega_{k,r} \mid r = 1, ..., R_k\} \in \Omega_k$. The unknown variables in (9) are e_ℓ, e_p, $\alpha_{k,\ell,p}$ and $\tau_{k,\ell,p}$. Furthermore, observe that the equation has nonlinear terms of the form $\alpha_{k,\ell,p}\, e^{j\omega_{k,r}\tau_{k,\ell,p}}\, e_p[m]$. It is linear if some of the variables are fixed.

2.3. Example

In this paper, we simulate the following scenario. We assume there are two transmitters (labeled as 1 and 2) and two sensors (labeled as 1 and 2) as shown in Fig. 1. Each transmitter is positioned at 100m or 400m from the sensors. Since the speed of light is 3×10^8m/s, the time delays between the transmitters and sensor 1 are $\tau_{1,1} = 0.33\mu$s, $\tau_{2,1} = 1.33\mu$s, and the time delays between the transmitters and sensor 2 are $\tau_{1,2} = 1.33\mu$s, $\tau_{2,2} = 0.33\mu$s.

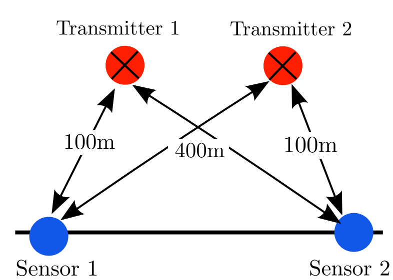

Fig. 1. Layout of the sensors and transmitters used in the simulation.

Relation (9) between the sensors requires the TDOA $\tau_{k,\ell,p}$ and the relative amplitude ratio $\alpha_{k,\ell,p}$. Using $\tau_{k,\ell,p} = \tau_{k,p} - \tau_{k,\ell}$, we find that $\tau_{1,1,2} = 1\mu$s and $\tau_{2,1,2} = -1\mu$s. Next, the factor $\alpha_{k,\ell,p}$ in (9) can be derived using the path loss equation (in dB):

$$\text{Path loss (dB)} = 10\, n \log_{10}(d) + C \quad (11)$$

where d is the distance between the sensor and transmitter, C is some constant and n is the path loss exponent. Assuming the signals travel in a free-space environment, we set $n = 2$. Hence, the relative path loss from one transmitter to the two sensors is $\pm$12dB. This means that $\alpha_{1,1,2}$ and $\alpha_{2,1,2}$ are approximately 4 and 0.25, respectively (since $\pm 20 \log_{10}(4) \approx \pm$12dB).

Next, we assume that the center frequencies of transmitters 1 and 2 are 512MHz and 536MHz, respectively, and both signals have a bandwidth of 20MHz. We also assume that the received signals are down-converted by a frequency shift of 500MHz and sampled at 100MHz. We also assume N=3200 samples are acquired in each realization. White Guassian noise with standard deviation of 0.001 is added to the data. Realizations of the ideal and distorted spectra at sensor 1 is shown in Fig. 2. For simplicity, we assume that the transmitted signals are bandlimited Gaussian signals.

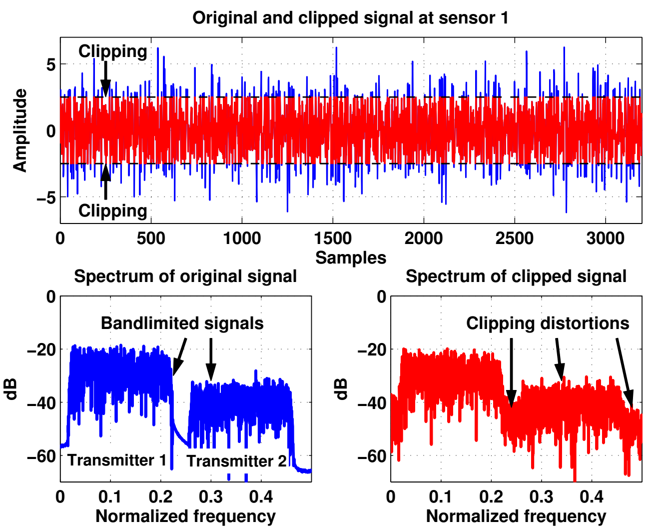

Fig. 2. The plots show both the time-domain and frequency-domain of the original and clipped signals at sensor 1. Details of the simulation parameters are in Section 2.3. The clipping level CL is $\pm$3.5.

3. PROPOSED SOLUTION

We propose a solution using data from two sensors and relation (9). Before describing the solution, we highlight two issues in (9) that relate to the nonlinearity of the problem and the detection of the frequency bands of the transmissions.

First, to convert problem (9) to a linear form, we have to fix some of the variables. For example, if we assume that the TDOA $\tau_{k,1,2}$ is fixed, it is possible to convert the equation to a linear form (refer to the next section). The maximum absolute bound on the TDOAs is the distance between the two sensors divided by the speed of light, which we denoted as τ_{max}. In this work, we uniformly discretize the possible TDOAs into a list that lies between $-\tau_{\text{max}}$ and τ_{max}. The quantity $\tau_{k,1,2}$ is estimated by scanning through the TDOA list and solving the optimization problem (12) for each TDOA. The estimated TDOA is the one that gives the minimum cost value.

$$\begin{aligned} \underset{\alpha_{k,1,2}}{\text{minimize}} \quad & \|[-Q_{k,1,2} x_2]\alpha_{k,1,2} + W_k x_1\|_2^2 \\ \text{subject to} \quad & \alpha_{k,1,2} \geq 0 \end{aligned} \tag{12}$$

where x_1 and x_2 are defined in (5). Initially, e_1 and e_2 in (5) are unknown. Therefore, they are set to 0 and (12) can be used to estimate $\tau_{k,1,2}$. The TDOA estimation can be repeated and improved, when e_1 and e_2 are estimated as shown in the next sections.

Moreover, the frequency bands of the transmissions, Ω_k (which are used to create W_k), are assumed to be known. If Ω_k is unknown, we assume it can be detected from the data under certain conditions. For example, if one of the sensors' data do not have clipping, we can use its data to estimate Ω_k. If all the sensors have clipped data, then we will need to estimate them from their distorted spectra. From [13, 14], we know that when clipping occurs, the resultant signal can be approximated by some linear combination of the original signal $x_\ell[n]$ and its higher odd powers of the signal $\{(x_\ell[n])^3, (x_\ell[n])^5, ...\}$. Usually, the power of the higher odd terms is smaller than the original signal. Therefore, in [13, 14], some of the transmission bands can be detected by finding the dominant frequency bands in the clipped data. If we have multiple sensors, we assume that by selecting the dominant frequency bands across the sensors, we will be able to estimate Ω_k. For example, in the distorted spectrum shown in Fig. 2, we can detect a transmission band around the normalized frequency of 0.12 (this is the signal from transmitter 1). Conversely, if we have access to the data of sensor 2, we can detect the signal from transmitter 2. Hence, we can obtain the entire transmission bands by combing the detections from the two sensors.

3.1. No clipping in one of the sensors in (9)

Let us consider the case where sensor 2 has no clipping. From (5), this means that e_2 is an empty vector and $x_2 = \check{x}_2$. The unknown variables that remain in (9) are e_1 and $\alpha_{k,1,2}$. Therefore,we can estimate them by solving the optimization problem:

$$\begin{aligned} \underset{e_1, \alpha_{k,1,2}}{\text{minimize}} \quad & \sum_{k=1}^{K} \left\| \begin{bmatrix} W_k B_1 & -Q_{k,1,2} x_2 \end{bmatrix} \begin{bmatrix} e_1 \\ \alpha_{k,1,2} \end{bmatrix} + W_k \check{x}_1 \right\|_2^2 \\ \text{subject to} \quad & e_1 \geq 0, \\ & \alpha_{k,1,2} \geq 0 \end{aligned} \tag{13}$$

3.2. Clipping in both sensors in (9)

Now, we consider the case when both sensors have clipping. We rewrite (9) as

$$\begin{aligned} W_k(\check{x}_1 + B_i e_2) &\approx \alpha_{k,1,2}\, Q_{k,1,2}(\check{x}_2 + B_2 e_2) \\ &\approx Q_{k,1,2}\check{x}_2 \alpha_{k,1,2} + Q_{k,1,2} B_2 e_2 \alpha_{k,1,2} \end{aligned} \tag{14}$$

where the variables are e_1, e_2 and $\alpha_{k,1,2}$.

Note that there is a nonlinear term involving the product of e_2 and $\alpha_{k,1,2}$. As such, we propose a suboptimal solution that might converge to a local minimum. This iterative solution fixes the variables e_2 and $\alpha_{k,1,2}$ alternatively. From (5), if we fix e_2, then x_2 is fixed. Therefore, we can use (13) to solve for $\alpha_{k,1,2}$ and e_1. Alternatively, if $\hat{\alpha}_{k,1,2}$ is estimated and fixed, we can find e_1 and e_2 by solving

$$\begin{aligned} \underset{e_1, e_2}{\text{minimize}} \quad & \sum_{k=1}^{K} \left\| P_{k,1,2} \begin{bmatrix} e_1 \\ e_2 \end{bmatrix} + W_k \check{x}_1 - Q_{k,1,2} \check{x}_2 \hat{\alpha}_{k,1,2} \right\|_2^2 \\ \text{subject to} \quad & e_1 \geq 0, \\ & e_2 \geq 0, \\ & P_{k,1,2} = \begin{bmatrix} W_k B_1 & -Q_{k,1,2} B_2 \hat{\alpha}_{k,1,2} \end{bmatrix} \end{aligned} \tag{15}$$

In summary, the proposed solution is:

Clipping level	2.5	3.0	3.5	4.0	4.5	5.0	5.5	6.0	6.5	7.0
Number of realizations with clippings	100	100	100	100	100	100	100	98	59	31
Average clippings[%]	17.5	10.2	5.6	2.9	1.4	0.6	0.3	0.1	0.06	0.04

Table 1. Number of realizations with clipping, and the average clipped samples as the clipping level is varied.

1. Initialize $\hat{\alpha}_{k,1,2}$, $\hat{e}_1$ and $\hat{e}_2$ to 0. Uniformly discretize between $\pm\tau_{\text{max}}$ to create a TDOA list.
2. Let $x_1 = \check{x}_1 + B_1\hat{e}_1$, $x_2 = \check{x}_2 + B_2\hat{e}_2$ and scan through the TDOA list using (12). Find the $\hat{\tau}_{k,1,2}$ from the TDOA list that gives the minimum cost value.
3. Use (13) to solve for $\hat{\alpha}_{k,1,2}$.
4. Use $\hat{\alpha}_{k,1,2}$ and (15) to solve for $\hat{e}_1$ and $\hat{e}_2$.
5. Repeat step 2 to 4 until the solution converge.

4. SIMULATIONS

We simulate the scenario described in Section 2.3. In the simulations, we vary the clipping level from 2.5 to 7.0 in steps of 0.5. The number of realizations where clipping occurs (out of 100 simulations) and the average percentage of clipped samples (out of N=3200) are shown in Table 1. The average power and standard deviation of the received signal is 3.37 and 1.84, respectively. The TDOA list used in estimating $\tau_{k,1,2}$ in (12) is between -1.7μs to 1.7μs in steps of 1ns.

For comparision purposes, we implement an algorithm that is similar to [5, 6], which only uses a single data stream (i.e., sensor 1's data). We did not compare with other works cited in Section 1 because their assumptions (e.g., 2 times oversampling or sparse representation) are not valid in these simulations, or their solutions use some approximation to model the clipping effects. The performance metric is evaluated using the signal-to-noise ratio (SNR):

$$\text{SNR} \triangleq 20\log_{10}\left(\frac{\|x_1\|}{\|x_1 - \hat{x}_1\|}\right) \quad (16)$$

4.1. Prior work [5, 6]

The prior works [5, 6] solve the clipping problem by minimizing the noise in the out-of-band frequency region. We implement this idea to compare with our proposed method. Recall from (7) that we define W_k to correspond to the transmission bands of the signals. Conversely, we can also define a matrix W^c that corresponds to the out-of-band frequency region. This matrix is simply the rows of the DFT matrix that do not belong to any of the transmission bands in $\{\Omega_1, ..., \Omega_K\}$. Hence, given W^c, we can remove the clipping distortions by finding e_1 that minimizes the noise in this frequency region. We denote this algorithm as Prior work 1:

$$\begin{aligned} &\underset{e_1}{\text{minimize}} \quad \|W^c B_1 e_1 + W^c \check{x}_1\|^2 \\ &\text{subject to} \quad e_1 \geq 0 \end{aligned} \quad (17)$$

4.2. Method 1: No clipping in sensor 2

For the first set of simulations, we consider the case when *only* sensor 1 experiences clipping, while sensor 2 does not. Therefore, we do not need (15) to recover the signal. Hence, when we use the proposed solution stated at the end of Section 3.2, we skip step 4. We denote this algorithm as Method 1. Figure 3 shows the SNR of the clipped signal before compensation and the SNR of the recovered signal using the Prior work 1 and Method 1.

4.3. Method 2: Clipping in both sensors

Next, we consider the case when *both* sensors experience clipping. Hence, we use the proposed solution stated at the end of Section 3.2. We denote this algorithm as Method 2. Figure 3 shows the SNR of the clipped signal before compensation and the SNR of the recovered signal using the Prior work 1 and Method 2.

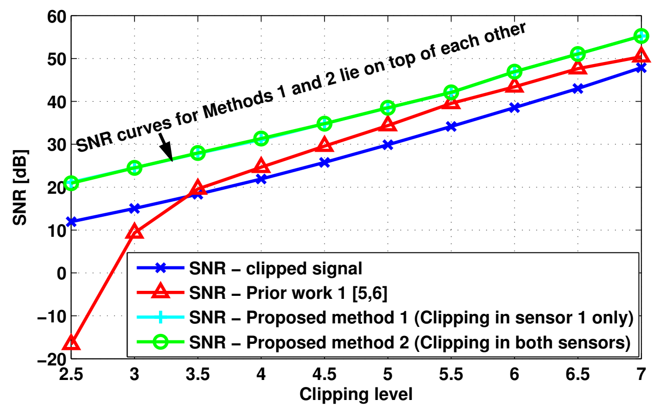

Fig. 3. The plot shows the SNR of the clipped signal and the recovered signal using Prior work 1 [5, 6], Methods 1 and 2. (Method 1 is used when only sensor 1's data has clipping distortion, and Method 2 is used when both sensors' data have clipping distortions.)

5. DISCUSSION AND CONCLUSION

The simulation results show that the proposed solution (Methods 1 and 2) work better than the prior works [5, 6], in the scenario where the data are not overly-sampled. The average SNR improvement using both Method 1 and Method 2 is 9dB. We also observed that when both sensors' data are clipped, we still obtain similar performance as the case when only 1 sesnor's data is clipped. We believe this is because (refer to Fig. 2) sensor 1 received a strong and weak signal from transmitters 1 and 2, respectively. Hence, the clipping distortions have a larger impact on the weak signal from transmitter 2. Whereas, in sensor 2, the effect is reversed; a strong signal from transmitter 2 is received, and the clipping distortions have less impact on the signal from transmitter 2. Therefore, the simulation shows similar performance when either one sensor or both sensors have clipping. We expect the performance for these two cases will change when, say, sensor 2 moves closer to sensor 1. There is a drop in performance in the prior work when the clipping level is low (2.5 to 3.0). This is because the number of clipping exceeds the rows of W^c in (17), and the linear system of equations become under-determined.

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