Sampling Jitter Estimation and Mitigation in Direct RF Sub-Sampling Receiver Architecture

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***Abstract*—The sampling jitter is particularly problematic in systems where high-frequency signals are sampled. This paper addresses the sampling jitter estimation and cancellation task in the direct RF sub-sampling type radio receivers. The proposed jitter estimation method is based on carefully injecting or superimposing an additional reference signal to the received signal at sampler input. Proper digital signal processing methods are then devised and applied to estimate the sampling jitter realizations from the obtained jittered samples. Using these jitter estimates, combined with proper jitter modelling, the jitter effects can then be efficiently removed from the actual received signal. Careful performance analysis of the overall estimation-cancellation scheme is also carried out using computer simulations with 3GPP LTE type multicarrier signals, assuming also different amounts of RF filtering prior to RF sub-sampling stage.**

***Keywords*—Sampling jitter; phase noise; mitigation; direct RF sampling; bandpass sampling; sub-sampling**

I. INTRODUCTION

Sampling of high-frequency signals, or signals with powerful interference at nearby frequencies, poses relatively high requirements for the timing accuracy of the sampling process [7]. In traditional communications receivers, down-conversion and analogue filtering are used to bring the signal down to lower frequencies and attenuate most of the interference near the interesting frequencies [5]. This results into relatively relaxed requirements for the dynamics and timing accuracy of the sampling circuitry. However, when emphasizing radio flexibility and re-configurability, more and more of the selectivity filtering is moved to digital domain. Similarly from the frequency-translations point of view, applying sampling already to higher-frequency signals is one of the main trends currently. Under these working assumptions, timing inaccuracies in the sampling process, called sampling jitter, become a severe problem. From the future wireless communications point of view, understanding, modelling and mitigation of the sampling jitter is thus a very interesting and important topic.

Sampling of very high-frequency signals with possibly powerful interferers at neighbouring frequencies is culminated in the so-called direct RF sampling (DRFS) receiver architecture. DRFS itself is, as the name implies, a receiver architecture in which the sampling of the incoming signal is done already at the radio frequencies (RF). DRFS minimizes the amount of analogue components in the receiver, thus emphasizing re-configurability and also potentially minimizing size, power consumption and costs of the receiver compared to more traditional radios. However, the DRFS concept still has many practical implementation issues to be solved [5], [11], and it is thus not commonly considered feasible especially for mobile terminal receivers with today's implementation technologies. The most notable problem with the DRFS approach is that it indeed poses very high demands for the quality of the sampling process. With current technologies, the combined requirements of relatively high sampling frequency and high resolution and timing accuracy for the used sampling circuitry result into relatively high power consumption, which in turn is one of the main concerns in mobile terminals. To circumvent this, digital signal processing (DSP) methods can be developed to lower the quality requirements for the sampling process.

In the literature, DSP-based estimation and mitigation of sampling jitter is not too widely researched because in most of today's receiver architectures, sampling takes place at fairly low frequencies. However, as indicated earlier, this might not be the case in near future, since minimizing the amount of analogue parts in receivers becomes more and more interesting. Furthermore, DSP capabilities of the mobile devices are continuously increasing. Recently, in [9], use of phase noise mitigation techniques [12] in mitigation of sampling jitter in bandpass sampling receivers was proposed for orthogonal frequency division multiplexing (OFDM) systems. In addition, the authors of [6] have proposed a technique to remove sampling jitter effects from general narrowband signals with help of a reference tone. In this paper, the idea of carefully designing and injecting a reference tone on top of the received signal at the sampler input signal is further developed. This paper first proposes an efficient technique to estimate the sampling jitter realizations with the help of such reference tone. The estimates of the jitter realizations are then used to mitigate the effects of the sampling jitter from the actual received waveform, without the limitations of narrowband signals and small jitter values as was assumed in [6]. Special emphasis in modelling and algorithm development is put to the DRFS receiver case where only partial selectivity is implemented at RF prior to sampling.

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The rest of the paper is organized as follows. In Section II, a novel method to estimate sampling jitter realizations is proposed utilizing a carefully injected reference tone. In addition, the Section discusses design considerations and limitations in applying the estimation principle in DRFS receiver context. Section III then proposes methods to use the acquired estimates of the sampling jitter realizations for actual mitigation of the jitter from the received signal. This covers both narrowband signal case and a more general DRFS receiver case. The performance simulations and the corresponding analysis are given in Section IV, whereas Section V finally concludes the work.

II. Estimating the Sampling Jitter Realization from the Sampled Reference Tone

In this Section, an efficient technique to estimate the sampling jitter realizations ($\hat{\zeta}_n$ in Fig. 1) from sampled signal is proposed. This is done by using the idea of injecting a reference tone to the signal prior to the sampling circuitry [6] combined with proper processing of the jittered samples.

A. Sampling Jitter Estimation at Principal Level

First, let $r(t)$ denote the incoming received signal, and the idea is to inject a reference tone, say $A_{ref}\cos(2\pi f_{ref}t)$ on top of it where f_{ref} denotes the reference tone frequency. Here only one sinusoidal reference signal is used, but also other kinds of known signals could in principle be used, if considered beneficial. Now, with suitable reference tone injected, the sampler input signal is of the form

$$x(t) = r(t) + A_{ref}\cos(2\pi f_{ref}t) . \tag{1}$$

Therefore, the jittered output of the sampling process is

$$\begin{aligned} x_n &= r(nT_s + \zeta_n) + A_{ref}\cos\left[2\pi f_{ref}(nT_s + \zeta_n)\right] \\ &= r(nT_s + \zeta_n) + \frac{A_{ref}}{2}\left(e^{j2\pi f_{ref}(nT_s+\zeta_n)} + e^{-j2\pi f_{ref}(nT_s+\zeta_n)}\right) \end{aligned}, \tag{2}$$

where T_s is the nominal sample interval, n is sample index and ζ_n is the sampling jitter at the n-th sample moment. Now, given that the reference frequency f_{ref} is outside the band of $r(t)$, we can use complex digital mixing and lowpass filtering (LPF) to isolate the jittered reference tone. Thus at the output of the digital mixer (running at f_{ref}) followed by LPF, we have

$$y_n = \mathrm{LPF}\left(x_n \times e^{-j2\pi f_{ref}nT_s}\right) \simeq e^{j2\pi f_{ref}\zeta_n}/2 . \tag{3}$$

This represents a sampled reference tone where jitter is seen essentially as phase noise. However, as already indicated above, the last equality is only approximately true because, due to jitter, the spectra of all the components in the sampled signal are spread around their original frequency contents. However, with practical sampling clocks, it is expected that there is some correlation between consecutive jitter values [4] and thus most of the energy of the sampled signal components is still located at the original frequencies. Now, based on (3), estimates for the sampling jitter values can be obtained as

$$\hat{\zeta}_n = \arg\{y_n\}/(2\pi f_{ref}) . \tag{4}$$

Like discussed already shortly above, there are some essential limitations in the estimation process which should be carefully understood. The reference frequency f_{ref} must be selected so that it is possible to sufficiently separate (filter) the reference tone from the actual incoming signal $r(t)$. In practice, the amount of this separation depends on (*i*) the frequency separation between $r(t)$ and f_{ref}, (*ii*) the amount of jitter and (*iii*) the power ratio between $r(t)$ and the reference tone. In the selection of f_{ref}, it should also be kept in mind that jitter has heavier effect on high-frequency signals, and thus the higher the selected reference frequency is, the easier it is to detect the phase behaviour of the isolated reference tone in (3)-(4). It should also be acknowledged that the reference signal itself must be very accurately known, and it is likely that the selection of f_{ref} affects also the generation accuracy of the reference.

Not only the frequency of the reference tone, but also its amplitude is important in the design. On one hand, if the dynamic range in the incoming signal is high, then also high number of bits is needed in the digital domain signal processing. On the other hand, the sampling jitter itself is easier to estimate if the reference tone is strong (compared to $r(t)$), so from the estimation point of view, we would like to use as powerful reference as possible. Thus a proper compromise is needed in dimensioning the reference signal power in practice.

B. Special Considerations for Sampling Jitter Estimation in DRFS Receiver Context

In applying the above jitter estimation principle in bandpass sub-sampling based DRFS receivers, all the above considerations are still valid. However, stemming from the use of sub-sampling, in which aliasing is used in a controlled manner to bring the incoming RF signal closer to baseband, some additional design considerations must also be taken into account. So in general, bandpass sub-sampling can introduce desired and non-desired aliasing. Thus proper combination of preliminary RF-filtering and choice of the sampling frequency is needed so that non-desired aliasing is minimized. In the proposed jitter injection and estimation technique, this must be done such that minimal aliasing happens over the reference tone, as the jitter estimation is quite sensitive to any additional interference. Of course in order to ease the RF-filter design, the aliasing pattern should be taken into account already when selecting the reference tone frequency. Notice that in general, with high sub-sampling ratio (ratio of the incoming RF-signal centre-frequency and the used sampling frequency), there is anyway lots of flexibility in designing the reference frequency.

III. Jitter Mitigation

This Section proposes an efficient way to use obtained jitter estimates in the jitter mitigation. First, for reference purposes, we study the jitter mitigation task in a similar case that was considered in [6]. Then in the second part of this Section, we focus on the jitter mitigation task in general high-frequency bandpass sampling receiver case.

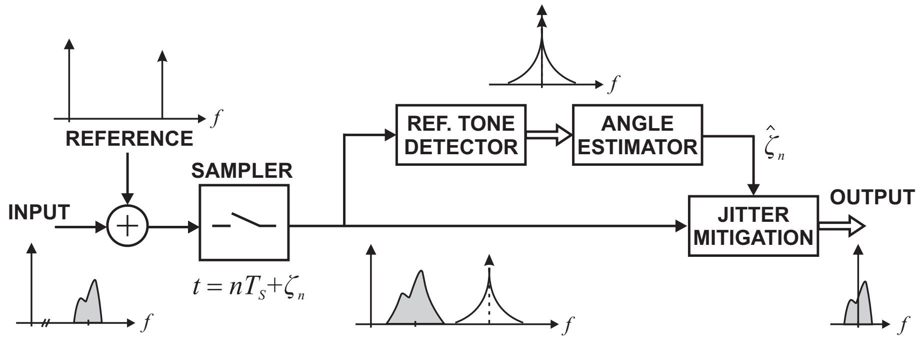

Fig. 1. General illustration of super-imposing a reference tone at sampler input which is then used for digital jitter estimation and cancellation.

A. *Case of Narrowband Signal with Nearby Interferer*

In [6], the case of narrowband target signal with very powerful interfering adjacent signal was considered. This case is somewhat impractical since huge power differences between the information signal and the neighbouring channel interference were assumed. Furthermore, only very narrowband information and adjacent channel signals were considered, and the jitter values were also assumed very small. Such case is, however, also considered in this Section, as a reference, to demonstrate that the jitter estimation and cancellation methods proposed here outperform the developments of [6] also under similar working conditions assumed in [6].

Thus here we consider a received signal of the form

$$x(t) = A_c \cos\left(2\pi f_c t + \theta_c\right) + A_i \cos\left(2\pi f_i t + \theta_i\right) \quad (5)$$

as the input of the sampler, where $r(t) = A_c \cos\left(2\pi f_c t + \theta_c\right)$ denotes the narrowband target signal, centred at f_c , and $A_i \cos\left(2\pi f_i t + \theta_i\right)$ models the adjacent channel signal at unknown neighbouring frequency f_i . Then, after jittered sampling, we have

$$\begin{aligned} x_n = x(nT_s + \zeta_n) = A_c \cos\left[2\pi f_c (nT_s + \zeta_n) + \theta_c\right] \\ + A_i \cos\left[2\pi f_i (nT_s + \zeta_n) + \theta_i\right] \end{aligned}. \quad (6)$$

For jitter mitigation, this real-valued signal is converted to complex (I/Q) signal by complex frequency translation with desired signal centre-frequency f_c . This yields essentially

$$\begin{aligned} z_n &= \mathrm{LPF}\left(x_n \times e^{-j2\pi f_c n T_s}\right) \\ &\simeq A_c e^{j\theta_c} e^{j2\pi f_c \zeta_n} + A_i e^{j\theta_i} e^{j2\pi (f_i - f_c) n T_s} e^{j2\pi f_i \zeta_n} \end{aligned}. \quad (7)$$

Now to actually mitigate the jitter effect, which is seen here basically as phase noise, this signal is multiplied by the complex exponential $\exp(-j2\pi f_c \hat{\zeta}_n)$ where $\hat{\zeta}_n$ denotes estimated jitter. With ideal jitter estimates, this would yield

$$z_n e^{-j2\pi f_c \zeta_n} = A_c e^{j\theta_c} + A_i e^{j\theta_i} e^{j2\pi (f_i - f_c) n T_s} e^{j2\pi (f_i - f_c)\zeta_n} . \quad (8)$$

Thus the jitter is fully removed from the target signal and also the jitter effect due to neighbouring channel is reduced, being now relative to $\exp(j2\pi(f_i - f_c)\zeta_n)$. Examples will be given in Section IV.

B. *Case of Direct RF Sub-Sampling Receiver Architecture*

Here we consider a more general case of receiving a bandpass signal $r(t)$ which, depending on the RF filtering, may contain several modulated signals at neighbouring carriers. This signal is first written here as $r(t) = s_I(t)\cos[2\pi f_c t] - s_Q(t)\sin[2\pi f_c t]$ where f_c denotes the formal centre-frequency of the *overall received band* and $s_I(t) + js_Q(t)$ is the corresponding baseband equivalent. The resulting sample sequence evaluated at sample instants $nT_s + \zeta_n$ is given by $r_n = r(nT_s + \zeta_n) = s_I(nT_s + \zeta_n)\cos[2\pi f_c(nT_s + \zeta_n)] - s_Q(nT_s + \zeta_n)\sin[2\pi f_c(nT_s + \zeta_n)]$. Taking now the sub-sampling principle [3], [5] into account, this equals to

$$\begin{aligned} r_n = s_I(nT_s + \zeta_n)\cos[2\pi f_{IF} nT_s + 2\pi f_c \zeta_n] \\ - s_Q(nT_s + \zeta_n)\sin[2\pi f_{IF} nT_s + 2\pi f_c \zeta_n] \end{aligned}, \quad (9)$$

where f_{IF} denotes the aliased centre-frequency. Applying then digital I/Q down-conversion from IF to zero frequency yields

$$\begin{aligned} z_n &= \mathrm{LPF}\left(r_n \times e^{-j2\pi f_{IF} n T_s}\right) \\ &= \left[s_I(nT_s + \zeta_n) + js_Q(nT_s + \zeta_n)\right] e^{j2\pi f_c \zeta_n} \\ &= s(nT_s + \zeta_n) e^{j2\pi f_c \zeta_n} \end{aligned}. \quad (10)$$

Now, if the original RF centre-frequency f_c is much higher than the corresponding signal bandwidth, the jitter contribution on the composite modulating I and Q components is much lower than on the carrier components [10]. Thus we can approximate (10) as

$$\begin{aligned} z_n &\approx [s_I(nT_s) + js_Q(nT_s)] e^{j2\pi f_c \zeta_n} \\ &= s(nT_s) e^{j2\pi f_c \zeta_n} \end{aligned}. \quad (11)$$

This basically means that the dominant jitter effect on the sub-sampled RF signal can be approximated as phase noise in the composite carriers of the original bandpass signal. [9]

Based on the above, jitter mitigation using $\hat{\zeta}_n$ can be carried out by multiplying the observed complex sample stream z_n with $\exp(-j2\pi f_c \hat{\zeta}_n)$. Assuming perfect jitter estimates, this fully removes the jitter stemming from the original bandpass nature of the sampled signal, i.e. $z_n \times e^{-j2\pi f_c \hat{\zeta}_n} \approx s_I(nT_s + \zeta_n) + js_Q(nT_s + \zeta_n) \approx s(nT_s)$.

IV. Simulation Environment, Results and Analysis

First, this Section briefly shows the jitter mitigation performance of the proposed technique in case of narrowband signal with nearby interferer, similar to the case studied in [6] by Rutten et al. Then, the used simulation model for the DRFS case is described and the corresponding simulation results are given and analyzed.

A. Case of Narrowband Signal with Nearby Interferer

As an example, we consider a case with narrowband information signal centred at 50 MHz and a powerful sinusoidal interferer at 51 MHz. A reference tone of 20 MHz is injected at sampler input and the sampling frequency is 200 MHz. 20 ps root-mean-square (RMS) jitter is assumed in the sampling (for more detailed statistical properties of the jitter and the sampling clock using phase locked loop, refer to [9] and [12]). Then estimation and mitigation of the jitter is carried out as explained in Sections II and III. The obtained mitigation performance is illustrated in Fig. 2. Clearly very good jitter mitigation is obtained, and also the proposed method outperforms the reference method in [6] by a few dB's.

As already mentioned, this case is as such rather impractical. As can be seen from Fig. 2, the power difference between the interferer and the information signal is originally around 60 dB, but still, the SNR due to jitter is even without mitigation around 50 dB. This is of course due to the fact that 20ps RMS jitter is very small *compared to* used signal frequencies in the order of 50 MHz. The mitigation techniques still anyway improve the SNR by roughly 30 dB, even though the initial SNR would already be sufficient for detection in any practical communication application.

B. Simulation Model for DRFS Receiver Architecture Case

For DRFS architecture case, 3GPP LTE –like system [1] is simulated to keep comparability with the previous work in [9]. OFDM waveform with 1024 subcarriers, of which 600 are active, is deployed with 16QAM subcarrier modulation and 15 kHz subcarrier spacing. Signal is transmitted at 2.6 GHz carrier frequency. After transmitter, this signal travels through either plain additive white Gaussian noise (AWGN) or extended ITU-R Vehicular A multipath [8] channel. When entering the receiver RF sub-sampling stage, both cases with and without neighbouring channel interferers are considered. In the case with adjacent channel interferers, three sinusoidal interferers at 7.495 MHz, 7.5 MHz and 7.505 MHz offset from the desired signal carrier frequency are considered. In addition, bandpass-noise interferer with approximately 4 MHz bandwidth at 9.5 MHz offset from the desired carrier frequency is also added to the other side of the signal as seen in Fig. 3. The interfering signals are considered to be approximately 40 dB above the interesting OFDM signal. Thus the total incoming signal bandwidth is around 19-20 MHz. Stemming from the centre-frequency of 2.6 GHz and the total bandwidth, we use sub-sampling frequency of 16×1024×15 kHz = 245.76 MHz. This aliases the RF signal from 2.6GHz centre-frequency to 11×245.76 MHz – 2600 MHz = 103.36 MHz IF frequency in a controlled manner. At sampler input,

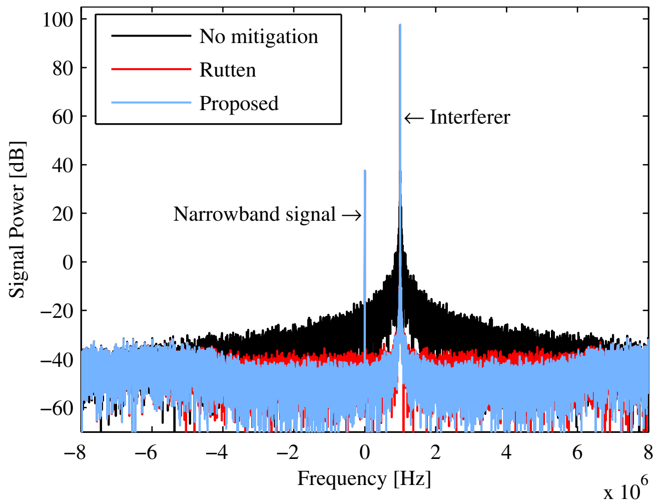

Fig. 2. Down-converted spectrum of signal environment, when 20 ps RMS jitter is applied to narrowband signal and its interferer.

the signal is injected with a reference tone of 36 MHz with approximately the same power level as the interferers. The aliased spectrum after sub-sampling is shown in Fig. 3 for both cases, namely interferer-free case and the case with adjacent channel interferers. After the sampling, the proposed jitter estimation and cancellation method is applied. Then the OFDM waveform is detected and the symbol error-rates (SER) are evaluated as performance indicators.

C. Simulation Results and Analysis

Simulation results for the DRFS receiver case with AWGN channel can be seen in Fig. 4 and Fig. 5. The results show that both techniques improve the receiver performance greatly compared to case without compensation. For interferer-free case, the proposed technique gives practically ideal performance over the whole evaluated SNR and jitter RMS regions. The reference technique from [6] performs nicely for lower than 5 ps jitter RMS. Its performance starts to decrease very fast after the 5 ps point. This is natural, because low-jitter assumption was made in derivation of the reference technique in [6]. In the more challenging case with the neighbouring channel interferers, the proposed technique performs again very well giving near-ideal performance with lower than 10 ps RMS jitter. With jitter RMS over 10 ps, some decrease in performance is seen, as the jitter-spread of the interferers gets higher and higher contaminating also the reference tone band. With the reference technique from [6], the performance is again limited to much lower jitter values.

The corresponding results for extended ITU-R Vehicular A multipath channel case are depicted in Fig. 6. The results are relatively similar to those given for the AWGN channel case. Of course, due to more challenging propagation environment, the detection error rates are generally higher. Thus in general, without mitigation, jitter begins to clearly affect the system performance only at somewhat higher jitter RMS values compared to the AWGN case with similar SNR. Anyway, the proposed technique can again efficiently push down the signal distortion up to 20ps RMS jitter range.

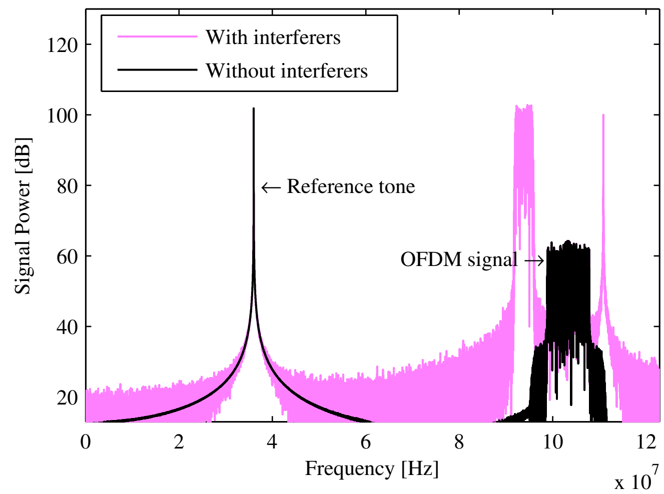

Fig. 3. Spectra of reference tone, OFDM signal (at 2.6 GHz originally) and interferers after aliasing when sampling at 245.76 MHz (20 ps RMS jitter).

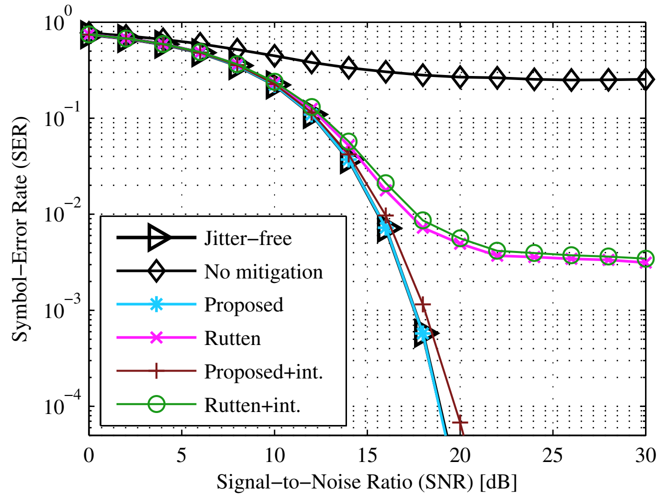

Fig. 5. SER as a function of SNR with fixed RMS jitter of 20 ps in AWGN channel.

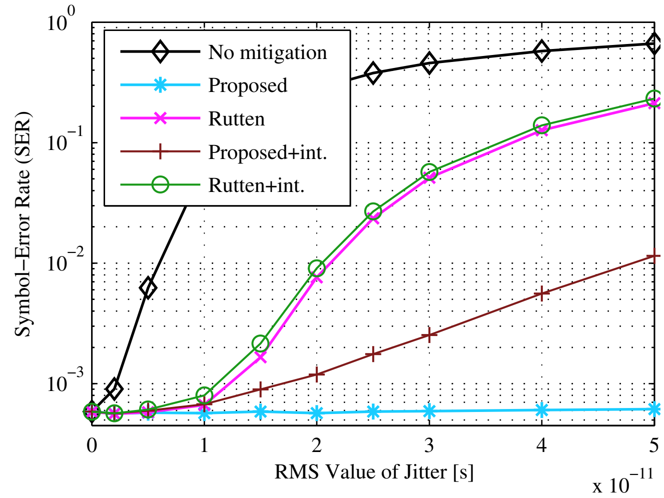

Fig. 4. SER as a function of sampling jitter RMS with fixed SNR of 18 dB in AWGN channel.

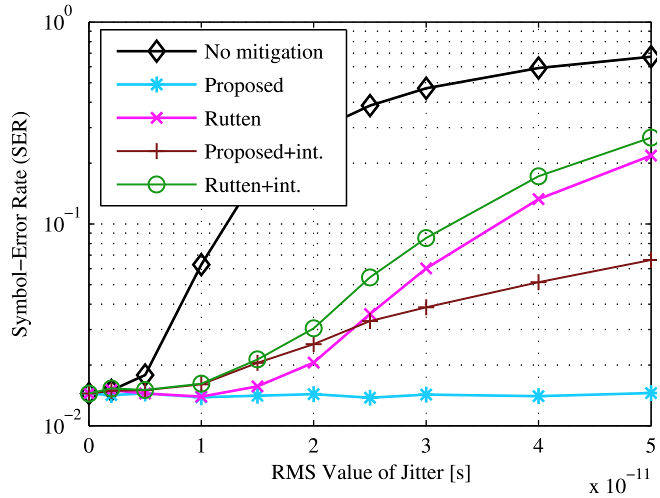

Fig. 6. SER as a function of sampling jitter RMS with fixed SNR of 26 dB in extended ITU-R Vehicular A multipath channel.

V. Conclusions

Sampling jitter is one of the main practical problems in direct RF sub-sampling receiver architecture. We proposed an efficient way to estimate the sampling jitter realization for every taken sample, with the help of an injected reference signal at the sampler input. We also proposed a way to use then these estimates to reduce the effects of the jitter, with special emphasis on DRFS receiver. Based on the performance simulations, the proposed estimation cancellation approach can efficiently mitigate sampling jitter also in the challenging case of strong neighbouring channels entering the sub-sampling radio.

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