A Digital Alias Cancellation Technique for Filtering-by-Aliasing Receivers

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There has been a growing trend of developing programmable transceivers, which are the key to realizing a true software-defined radio [1]. Some prior works, such as N-path filters and mixer-first receivers, utilized periodically time-varying (PTV) circuits and have shown some promises and achieved what conventional time-invariant circuits are incapable of. Among them, the filtering-by-aliasing (FA) receivers [2], [3] have demonstrated good potential by showcasing sharp bandpass filters with good linearity and programmability.

Shown in Fig. 1(a), in an FA receiver front-end, the time-invariant resistor in an *RC* integrator is replaced by a PTV one. Together with a baseband integrate-and-dump circuit and a passive mixer, the impulse response seen at the sampled output, $g(\tau)$, is controlled by the PTV resistance variation as shown in Fig. 1(a), which can be designed to realize very sharp analog FIR filters, while the carrier frequency is set by the LO. However, the achievable stopband rejection is finite. Unlike the mixer-first ones, due to the inherent sampling nature of the circuit, further filtering is not possible by cascading more baseband stages after an FA receiver [3]. This is because the blockers, *albeit* being significantly filtered, will fold in band after the sampler, as illustrated in Fig. 1(b) where the blue curve shows an example filter shape. The in-band signal and residual blocker aliases are no longer separable (here we use tonal signals for easy illustration, so they appear separable, but real-world signals are wideband), which potentially limits the SNR or even causes the signal to be buried by the blocker alias if the blocker is too strong. This is a fundamental limitation of the FA receivers. In this paper, we present an alias cancellation technique based on digital signal processing following the sampling operation and digitization, which effectively enhances the stopband rejection of the FA receivers.

The concept of the proposed DSP-based alias cancellation technique is shown in Fig. 2(a). The main-path FA receiver captures the desired signal and rejects unwanted blockers. The blocker in red will be significantly filtered, but still may hurt the SNR as can be seen at the output of the main-path ADC. An auxiliary receiver path is added in parallel to capture the blocker. Its LO, f_{aux}, is tuned such that the blocker f_{blk} at is in-band for this channel and will alias to the same baseband frequency as that in the main path. By properly scaling the blocker in the auxiliary path with a digital filter, $H(z)$, we can generate an exact replica of the main-path blocker's residual alias. Then, by subtracting the outputs of the two paths, only the desired signal will remain. Fig. 2(b) depicts the implementation of $H(z)$. $I_{\mathrm{main,aux}}$ and $Q_{\mathrm{main,aux}}$ are the I/Q outputs of the main and auxiliary receivers, respectively, and those labeled with a tilde are the corresponding digitized signals. Here, $H(z)$ is composed of two separate filters, one for the blocker's alias itself [$H_1(z)$], and one for the image of that [$H_2(z)$]. The image is due to mixing real wanted signals with complex LOs and will locate at the opposite frequency of the blocker's alias. A delay element, $H_{\mathrm{delay}}(z)$, is added at the output of the main path to equalize the delay difference in the two paths but provides no filtering. The complex conjugation operator is simple in real implementation, as it is nothing but adding a negative sign for the Q-channel output.

In principle, the auxiliary receiver can be of any type that can have its LO tuned to near an arbitrary blocker frequency. It also needs to be reasonably linear so that it will not be saturated by the blocker. Here, we demonstrate the proposed technique using another FA receiver channel and the chip in [3] is used, which inherently has two channels and is very linear. Fig. 3(a) shows the measurement setup, where a four-channel off-the-shelf ADC (AD9253 from Analog Devices) is used to digitize the outputs (four channels are needed because each receiver has I/Q outputs), and the DSP algorithm is implemented in MATLAB. Fig. 3(b) shows an example of the output spectra of the two-channel FA receiver captured by the ADC, at f_{LO} = 270 MHz, f_{aux} = 350 MHz, and f_{s} = 10 MHz with a sinusoid at 347.75 MHz sent to the input of the receiver. In this paper, for demonstration purpose and easy comparison, all measurements were performed using this setup of f_{LO}, f_{aux}, and f_{s}. The ADC ran at 60 MHz, higher than f_{s}, which was unnecessary and only used for easier measurement. The digital outputs were therefore decimated. The decimation filter was slightly over-designed so the components near $\pm f_{\mathrm{s}}/2$ is filtered out, but this is not required. As is apparent, the auxiliary channel captured the blocker while two residual aliases showed up at the output of the main channel, though filtered. The images and spurs, as well as out-of-band signals and nonlinear distortions, in the auxiliary path do not nominally matter because $|H(z)| << 1$. In fact, $|H(z)|$ is at the same level of the stopband rejection of FA, which was about –50 dB in this case.

The equalizing filter, $H(z)$, needs to match the relative frequency response between the two channels. For reasonable accuracy of $H(z)$, a 20-tap length FIR filter was chosen, which was sufficient for good cancellation according to simulation. Due to parasitics and other non-idealities, it is not practical to precisely estimate $H(z)$ analytically or via simulation. Instead, the filter coefficients were determined by making single-tone measurements during a foreground calibration period with the input frequency swept within the auxiliary channel's BW. We chose the number of tones swept to be equal to the number of taps, but this is not required. While background calibration of the filter responses or using non-single-tone calibration signals may also be possible, it is beyond the scope of this work. The measured relative frequency responses can be found by

$$\tilde{H}_1(j\Omega) = \mathcal{F}(\tilde{I}_{\mathrm{main}} + j\tilde{Q}_{\mathrm{main}})(j\Omega)/\mathcal{F}(\tilde{I}_{\mathrm{aux}} + j\tilde{Q}_{\mathrm{aux}})(j\Omega),$$
$$\tilde{H}_2(-j\Omega) = \mathcal{F}(\tilde{I}_{\mathrm{main}} + j\tilde{Q}_{\mathrm{main}})(-j\Omega)/\mathcal{F}(\tilde{I}_{\mathrm{aux}} - j\tilde{Q}_{\mathrm{aux}})(-j\Omega), \quad (1)$$

where Ω is the corresponding baseband frequency and $\mathcal{F}$ is the fast Fourier transform (FFT) operator. After finding $\tilde{H}_1$ and $\tilde{H}_2$, the corresponding digital filter coefficients, $h_1[n]$ and $h_2[n]$, can be obtained by performing inverse FFT on $\tilde{H}_1$ and $\tilde{H}_2$. Fig. 4 shows the measured $\tilde{H}_1$ and $\tilde{H}_2$ and the corresponding frequency responses of $h_1[n]$ and $h_2[n]$ [i.e., $H_1(j\Omega)$ and $H_2(j\Omega)$]. The frequency responses match reasonably well except at the band edges. Around dc, the frequency responses are slightly altered because the input impedances at LO frequencies are slightly different [3]. Running at baseband rate f_{s} (10 MHz in our case), such 20-tap digital filters consume very little power and have little area overhead.

After applying the foreground calibration, to verify the effectiveness of the proposed technique, single-tone signals were sent to the receivers, and we look at the output spectra. Fig. 5(a) shows the output spectra with and without alias cancellation technique for an input frequency of 348.5 MHz. It is clear that the residual blocker aliases were further suppressed by 18 dB. Fig. 5(b) shows the effective stopband rejection for blockers within the auxiliary channel's band with and without the technique. On average, an improvement of 14 dB was observed. The test frequencies were intentionally chosen to be different from the calibration frequencies used to generate $h_1[n]$ and $h_2[n]$.

In addition to the tonal tests, we applied the technique on wideband blockers. Fig. 6(a) shows the output spectra with and without the technique for a wideband phase-modulated (PM) blocker centered at 347.5 MHz. The maximum aliases were suppressed by another 13 dB. The integrity of the desired signal was also verified, shown in Fig. 6(b), where a –5-dBm PM blocker centered at 347.5 MHz was accompanied by a –55-dBm in-band sinusoidal wanted signal. Apparently, the aliases were suppressed by 11 dB while the wanted signal is intact, hence improving the SNR. The magnitude difference of the desired signal before and after cancellation was less than 0.5 dB. The image of the signal itself was still present and canceled as it is outside the scope of this work. The hardware overhead includes 120 multipliers, 118 adders, and 120 delay cells, running at f_{s}. The proposed technique helps enhance the effective stopband rejection of FA receivers with the auxiliary channel's BW by ~14 dB. The residual alias problem is hence greatly mitigated, while without the proposed technique, once the blockers alias in-band, they cannot be further suppressed.

References:

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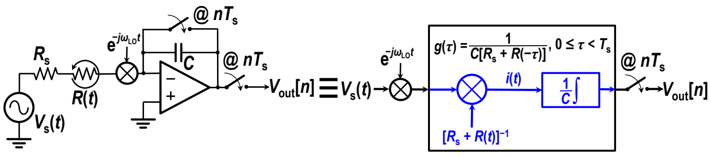
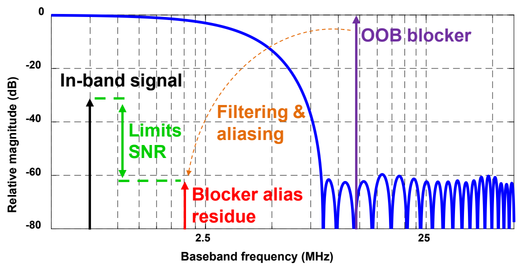
[2] S. Hameed and S. Pamarti, "Design and analysis of a programmable receiver front end based on baseband analog-FIR filtering using an LPTV resistor," *IEEE J. Solid-State Circuits*, vol. 53, no. 6, pp. 1592–1606, Jun. 2018.

[3] S. Bu and S. Pamarti, "A dual-channel high-linearity filtering-by-aliasing receiver front-end supporting carrier aggregation", *IEEE J. Solid-State Circuits*, in press.

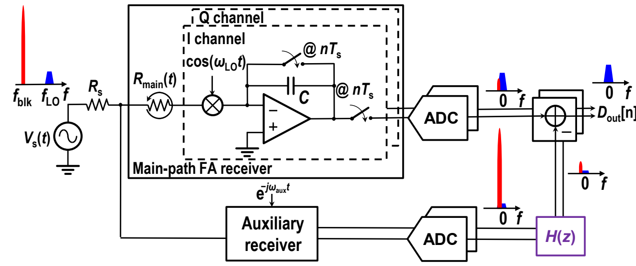


(a)



(b)

Fig. 1. (a) Block diagram of a filtering-by-aliasing (FA) receiver front-end [2] and (b) the attendant residual aliasing problem, illustrated using the baseband filter, i.e., omitting the mixer.



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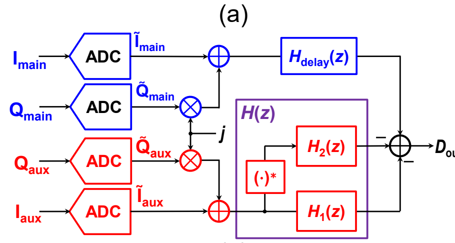
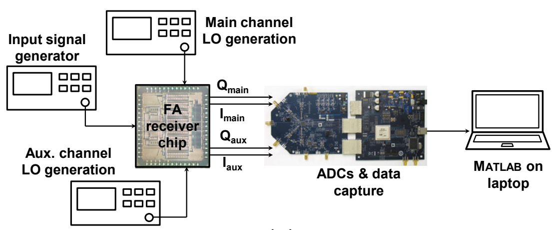


(b)

Fig. 2. (a) Concept of the proposed alias cancellation technique (each receiver has I/Q outputs) and (b) system-level overview of the implementation of the DSP. $(\cdot)^*$ is the complex conjugation operator.



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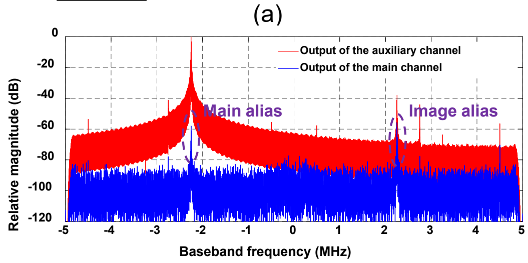
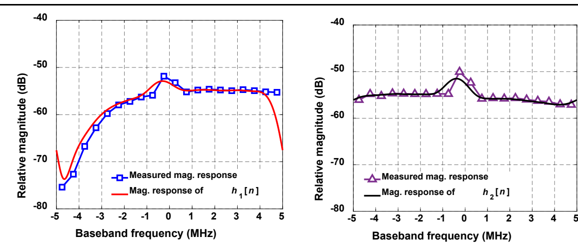
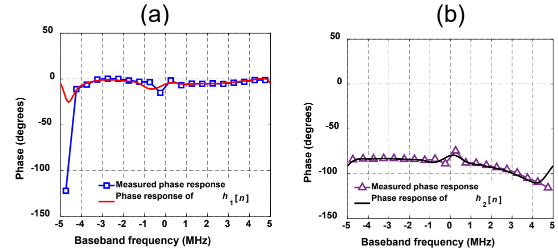


(b)

Fig. 3. (a) Experiment setup using the dual-channel FA receiver chip in [3] and off-the-shelf ADCs, and (b) example of the two channels' output spectra, where the main and auxiliary channels capture the blocker aliases and the downconverted blocker signal, respectively.

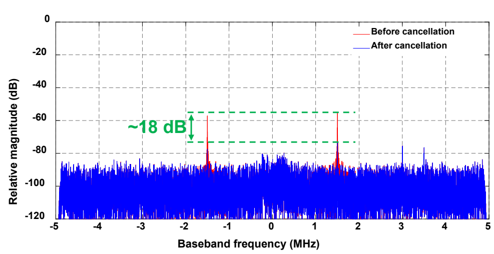


(a) (b)

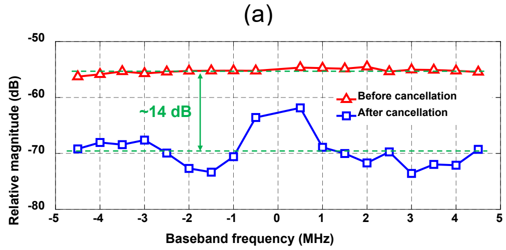


(c) (d)

Fig. 4. Comparison between the relative frequency responses of the measured results and digital equalizing filters (after removing artificial delay) in terms of magnitudes of (a) main and (b) image aliases, and phases of (c) main and (d) image aliases.

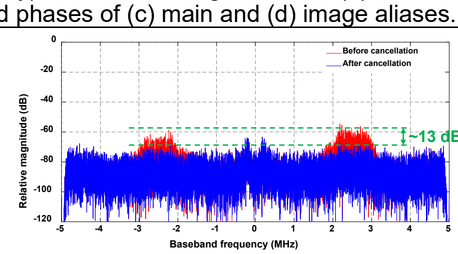


(a)

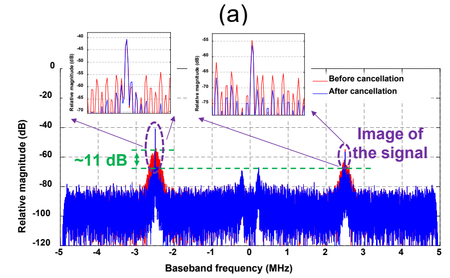


(b)

Fig. 5. (a) Baseband spectrum showing the effectiveness for a tonal blocker and (b) effective rejection of the blocker before and after cancellation across the band of interest.



(a)



(b)

Fig. 6. Baseband spectra showing cancellation for (a) wideband blocker and (b) wideband blocker accompanied by a tonal desired signal (with an input power difference of 50 dB).