Background
Some inaccuracies crept into the article regarding the operation and performance of the Focusrite Scarlett 2i2 2nd generation. The manufacturer send us some corrections and clarifications which we are happy to provide below. The highlighted red sections are the sections the comments apply to. – Ed.

1. Conversion Quality

Vanderkooy states (page 162, 1st paragraph under "System Architecture"):

There is some flexibility in how we choose to use the DACs and ADCs of a soundcard. We need only one DAC output channel for the system excitation, the second channel is not usually necessary, although it could have a polarity-inverted signal for driving balanced inputs, even though many devices already use balanced outputs. The two ADC input channels need antialiasing protection, and that is already provided by most stereo codecs in the soundcard. A soundcard capable of record/play at 96 or even 192 kHz is useful for a measurement system, since it is capable of measuring the response well beyond audible frequencies, allowing aspects like filter roll-off and out-of-band spuria to be studied. However, you must check if the soundcard actually delivers the increased bandwidth.

The Focusrite Scarlett 2i2 2nd-gen works nicely up to 40 kHz with sampling frequencies of 88.2 and 96 kHz, but at higher sampling rates there are relatively gentle filters that remove much of the higher frequencies, don't offer as much antialiasing protection, and have increased distortion. At 192 kHz the transfer function is 7dB down at 80 kHz. For audio we do not recommend 192 kHz for any soundcard. The ADCs and DACs are significantly compromised at these frequencies, and it is sometimes true even at 96kHz.

Some may consider this better for high-quality audio, but there is really no evidence for that. A simple 44.1/48 kHz soundcard is fine for normal work. For a measurement system, though, a higher sampling frequency may be useful.

Whilst we accept there is compromised performance at 192kHz with higher gain settings, there is no justification for most of these comments, especially as the author is not privy to the electronic design of the product. Because we use over-sampling ADC's and DAC's with a suitable analogue low-order low pass filter at their inputs and outputs, there is no compromise at higher sample rates. The anti-aliasing is provided by the decimation filters integrated within the converter integrated circuit (IC), as detailed in the article (page 161, "The Hardware: DACs and ADCS"); reference can be made to the Cirrus CS4272 data-sheet. We accept that, at 192k, we have made compromises in the analogue electronics that manifest as reduced bandwidth from the ideal, but we consider that a valid decision in our mission of providing the user with the best possible feature and performance set at the product price point. In this respect, the Scarlett 2i2 is the leader amongst its competition.

Additionally, the last sentence highlighted:

The ADCs and DACs are significantly compromised at these frequencies, and that is sometimes true even at 96 kHz.

This may be the case for soundcards in general (due to compromised design) but here the context of the text is inferring it is specific to the Scarlett 2i2, which is not the case. Therefore, this statement is inappropriate in the context of this paragraph; it would have been better placed prior to these comments, before the specific mention of the Scarlett 2i2.

We would prefer the highlighted section is re-phrased as below:

"The Focusrite Scarlett 2i2 2nd-gen works nicely up to 40 kHz with sampling frequencies of 88.2 and 96 kHz, but at higher sampling rates there is increased distortion, because of the higher bandwidth available for both harmonic content and noise. At 192 kHz and high gain settings the transfer function is 7dB down at 80 kHz (roughly 5dB on the input and 2dB on the output)! Thus, we do not recommend 192 kHz for any soundcard intended to be used for this purpose, although this performance would be perfectly acceptable in real recording situations that demand high bandwidth audio capture."

2. Phase Inversion

Vanderkooy states (page 172, 4th paragraph under "Characterizing Your Soundcard"):

You will have to measure your own soundcard, but we have measured a few popular ones for Windows 7 using the unbalanced output (half of the output if it's balanced). For the Behringer UCA202/222, SDAC=−1.22V/#, and SADC=−1.44V/#. For the ART USB Dual Pre, SDAC=+1.05V/# and SADC=+1.49V/#, using TRS inputs with potentiometer set to minimum. We tried 5 or 6 Dual Pre's and the values can differ by 10% between them. The soundcards were measured under Windows 7 with the recording gain set to 4, which is the recommendation in the brochure and suggested on the web. At a record gain of 3, the ADC can saturate before reaching digital full scale, not a recommended setting.

A Focusrite Scarlett 2i2 had SDAC=−1.59V/#, and SADC=+1.16V/# for the line inputs at 12:00 o'clock pot setting. Minimum gain is −23dB and maximum gain is +27dB, for a total gain range of 50dB. The instrument input has 9dB more gain than the line input setting, and the XLR mic input has 10dB gain above that. This otherwise superb soundcard has an inverting monitor output! Thankfully the ADCs are not polarity inverting, so your recorded music will be OK, unless the mics invert.

This is factually incorrect, verified by our own testing, also conceded by email by Vanderkooy. We would like the article to be corrected in line with the paragraph below:

"You will have to measure your own soundcard, but we have measured a few popular ones for Windows 7 using the unbalanced output (half of the output if it's balanced). For the Behringer UCA202/222, SDAC=−1.22V/#, and SADC=−1.44V/#. For the ART USB Dual Pre, SDAC=+1.05V/# and SADC=+1.49V/#, using TRS inputs with potentiometer set to minimum. We tried 5 or 6 Dual Pre's and the values can differ by 10% between them. The soundcards were measured under Windows 7 with the recording gain set to 4, which is the recommendation in the brochure and suggested on the web. At a record gain of 3, the ADC can saturate before reaching digital full scale, not a recommended setting. A Focusrite Scarlett 2i2 had SDAC=+1.59V/#, and SADC=+1.16V/# for the line inputs at 12:00 o'clock pot setting. Minimum gain is −23dB and maximum gain is +27dB, for a total gain range of 50dB. The instrument input has 9dB more gain than the line input setting, and the XLR mic input has 10dB gain above that."

3. Sweep Source Code

Phase Inversion

In the source code file *logsweep1rt.m* line 39 refers to the incorrect phase inversion within the Scarlett 2i2:

% S_dac=-1.59;% Focusrite 2i2 Yes, it inverts its monitor output!

This must be corrected to represent the phase correct functionality of the Scarlett 2i2:

% S_dac=+1.59;% Focusrite 2i2 No phase inversion