

A review of Marcel's solid state Firdac D/A converter.

Marcel, after having designed a very sophisticated Tube Dac, published his fully solid state Dac based on the same software driven FPGA, and since tubes are not my world, I became highly interested in this solid state design after having read several very positive reviews of the Tube Dac.

From Trenz in Germany I ordered the required LX45 FPGA to be paid in advance.

This was august 2021 and supply was projected for October that same year.

Before that date, I received information that delivery was going to be delayed for at least 6 months, and as it turned out later, even that was not going to be met.

So because working on a different project I decided to postpone the whole project and cancelled my order mid-2022.

Living in the same country as Marcel, I recently found him so generous to lend me his prototype, making it way easier for me to decide whether to reactivate the project or to leave it.

First impressions

First thing I did was listening to the Dac and I was pleasantly surprised.

It was the most analogue Dac I ever auditioned, that is to say in the pwm8 mode.

The other two modes, pwm4 and chaos, at least on my ESL based audio system, sounded somewhat less involving, so I concentrated mainly on this pwm8 mode.

However, there was something peculiar, small vocal groups and Jazz groups sounded fantastic in their restricted podium size, better than I heard ever before. But when playing classical music, it sounded like the whole orchestra was cramped on the same small podium and in a smaller concert hall.

So, I decided to take a closer look by measuring things, possibly shedding some light on the why.

And since I could hardly believe that digital part was causing a restricted sound stage, my main attention went to the reconstruction filter.

Preparations for setting up the measuring chain

My first thoughts were that the shared digital and analogue Signal Grounds could possibly be the cause by polluting the analogue signal with digital noise, but in a later stage this turned out difficult to prove in a measurable way.

I normalised my measuring system making that a 0dB sine, resulting in 1.925 Volt rms at the diff output, was displayed as 0dBV on my scope's screen.

In that way, in its most sensitive setting a -60dB input signal was being displayed as 1mV rms and the noise floor as displayed by the scope could be easily used to calculate the Dac's S/N.

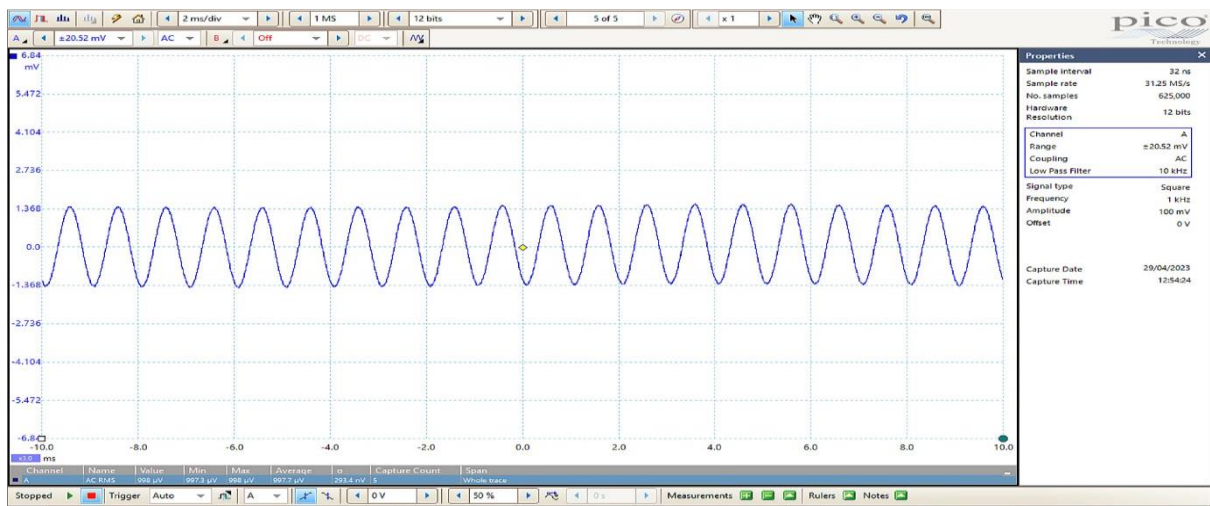


Fig 1, 1KHz input signal at -60dB in pwm8 coming from the Dac after normalisation.

Checking the same in the frequency domain gave this image.

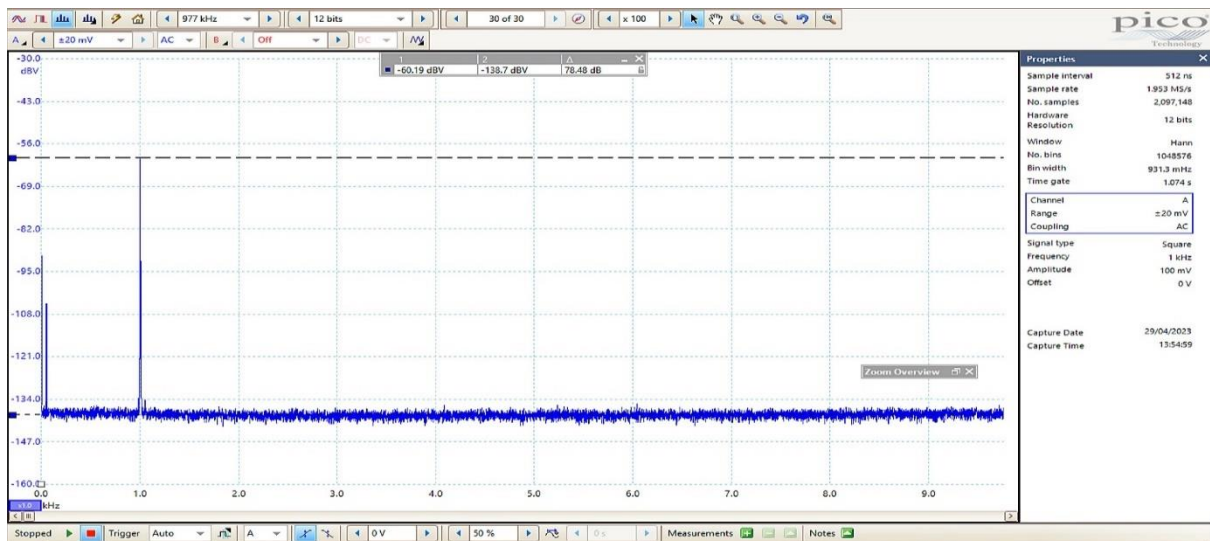


Fig 2, Spectrum from -60dB input signal with Hann Window.

A 60dB gain 5nV/rtHz preamplifier having a 1Mhz BW was used for the purpose, and to prevent any aliasing, the Logo on the right in the scope's above image reveals that the spectrum was recorded with a 1Mhz FFT .

With 1Meg bins, this gave a convenient bin width of ca. 1Hz.

The image in fig 2 was taken while using a Hann window to keep the 1Khz spectral line as narrow as possible.

However a Hann window will let the noise increase by 1.7dB, so for the coming precise noise measurements I used a rectangular window.

A complication for measuring the very low level signals was the large out of band noise created by the SDM. This would force me to set the scope on a lower sensitivity to prevent overloading. That's why an additional 30Khz filter was placed between 60dB measuring amp and oscilloscope. This enabled a higher sensitivity setting, thereby lowering the measuring chain's noise contribution to below -160dB/rtHz, a safe distance from the Dac's noise production, see Fig 3 below.

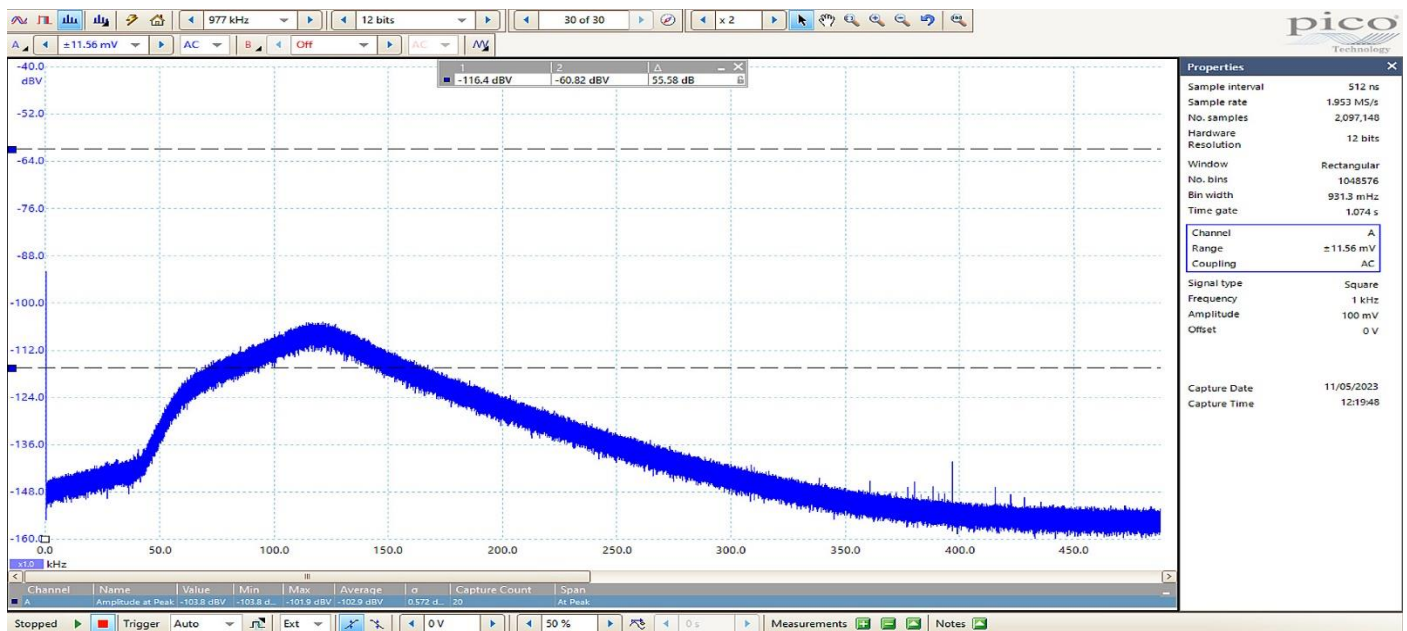


Fig 3, Noised spectrum in pwm8 mode, showing SDM noise behind the reconstruction filter.

Initially a non-stabilised 15V power supply was used, which resulted in AM modulating the LT3042 with 50Hz sidebands around the test tone.

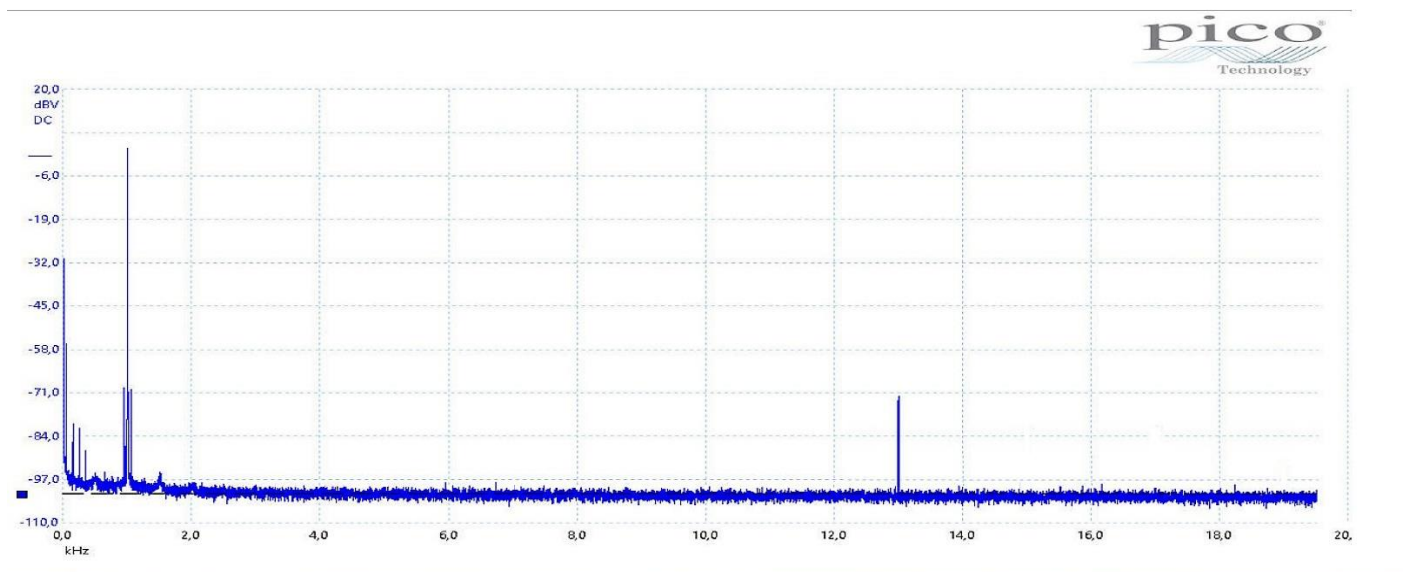


Fig 4, Spectral lines around 1Khz@0dB with a non-stabilised 15Volt supply, causing 50Hz AM modulation

That's why for this tests, a power supply equipped with LM317 type of regulators was used, which seemed o.k., see Fig 5 below, but for a definitive version, low noise LDO's would be advised to radically suppress 50Hz AM modulating from the critical 5Volt reference voltage.

For the LT3042 a ceramic cap on the set pin turned out to be the wrong choice, probably because of piezoelectric effect leading to producing strange harmonics, see the 13Khz spectral line in Fig 4 and Fig 5. Replacing this cap with an electrolytic alu cap radically solved the problem for ever.

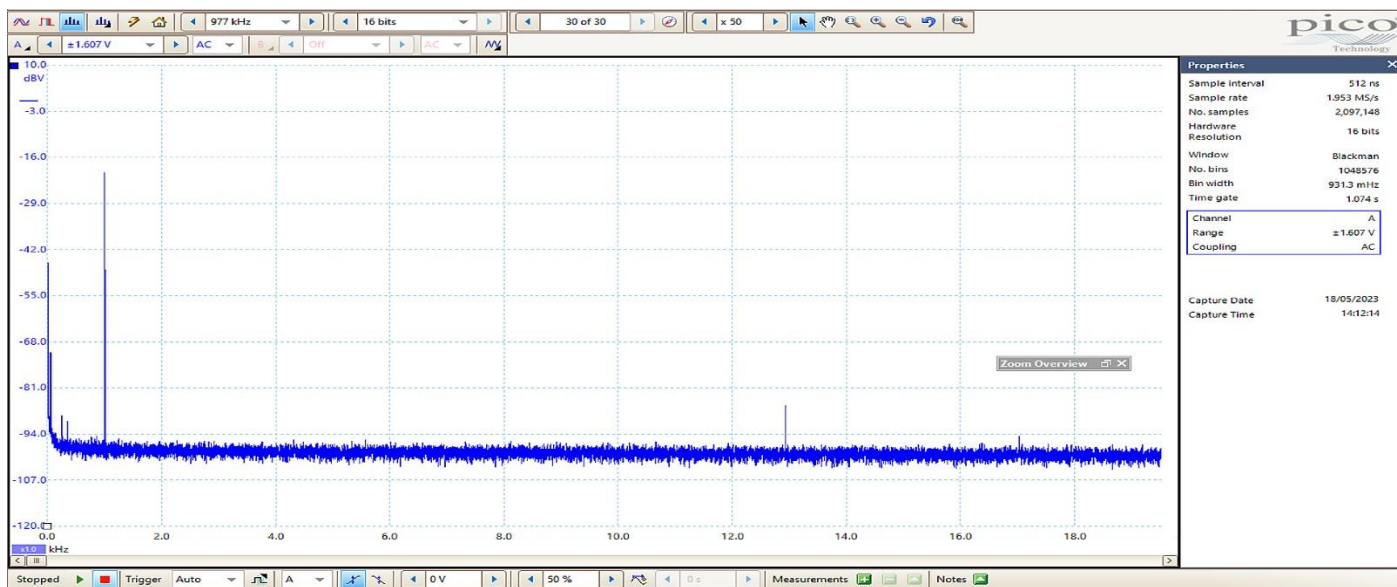


Fig 5, 13Khz spectral line caused by ceramic cap on LT3042, input -20dB.

Measuring the S/N and modifying the reconstruction filter to further improve S/N.

The first noise measurements, taking all the above into account, using a silent track filled with zeros with no jitter, I found a noise level of -145.1dB/rtHz

Disabling the SDM by leaving SW1 open, thereby forcing the Firdac output at a stationary level, I measured -145.9dB/rtHz.

The latter figure represents the noise from the reconstruction filter in isolation, obviously worsening the overall noise to the measured -145.1dB/rtHz.

However, even using this figure for a noise BW of 20Khz, which means adding $10 \cdot \log(20.000) = 43.0\text{dB}$ gives already a nice S/N of -102.1dB or -103.9 dBA, almost 6dB better than the -98.3 dBA that Marcel had previously published for pwm8.

So his Dac is even better than was thought !

But the fact that the reconstruction filter was rather noisy, made me simulate the filter in LTSpice which gave me almost exactly the same noise output, so I was on the right track to find a possible improvement.

Since the second and the third stage were contributing most noise, I changed the first stage into giving 4 times more gain, and at the end of the filter I divided the output by 4 to get the same original gain. At the same time the FR was kept intact up to 80Khz.

LTSpice thereby showed a large noise reduction of 5.5dB with this relatively simple alteration.

So by changing just a few components I now got the following noise readings, -147.9dB/rtHz for a silent track in pwm8 and -151.5dB/rtHz for the reconstruction filter with the SDM in stationary state, corresponding remarkably well with my LTSpice sim prediction. S/N now became $-147.9 + 43.0 = 104.9\text{dB}$ or 106.7dBA, a 2.8dB improvement over the original version.

But I wasn't yet satisfied with the above figures, because it was easy to calculate that the reconstruction filter was still worsening the Dac's output noise by some 2.5dB, and for such an excellent digital design it could be beneficial from a sound reproduction perspective, to go as low as possible for an overall balanced S/N between digital and analogue.

So I decided to change all components around reconstruction filter's stage 1 by a factor 4.6, expecting to reducing the overall filter's noise by another 2.6dB.

This implied reducing the Firdac resistors from 32k4 into 6k98.

Below is the circuit diagram showing how the complete filter became after that, including the 4 times larger gain for stage one and bringing gain back to its original value right before the output buffer.

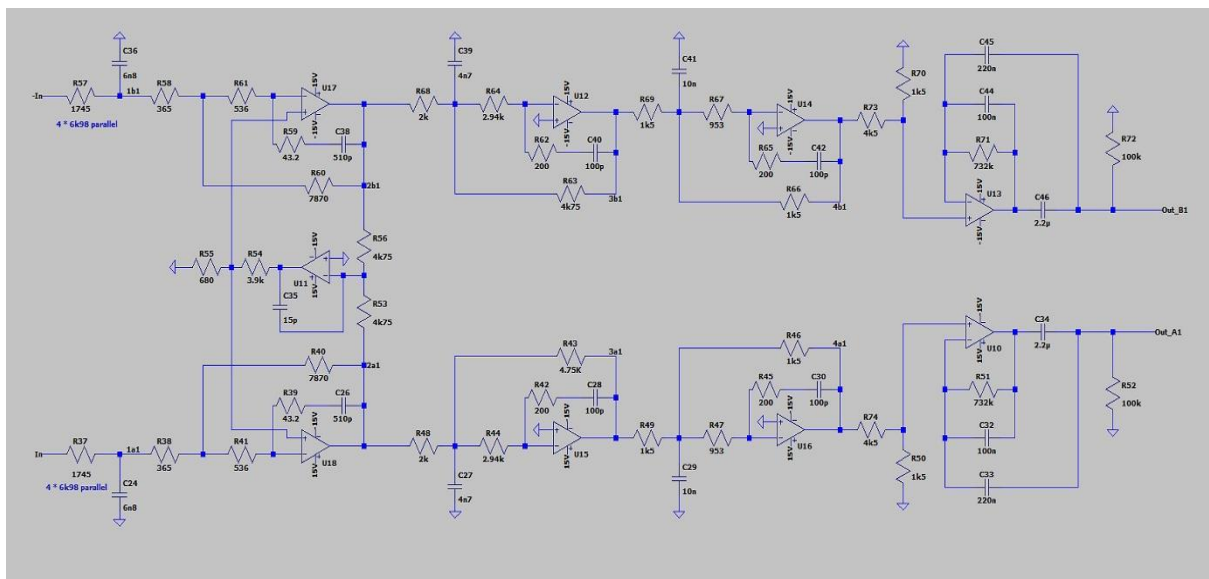


Fig 6, Reconstruction Filter with almost 8dB reduced noise over the original version.

After having again carefully adjusted the scope's gain to get the 1mV rms reading on the oscilloscope with -60dB input signal as in Fig1, 0dB now giving 1.967 Volt rms, measurements could start.

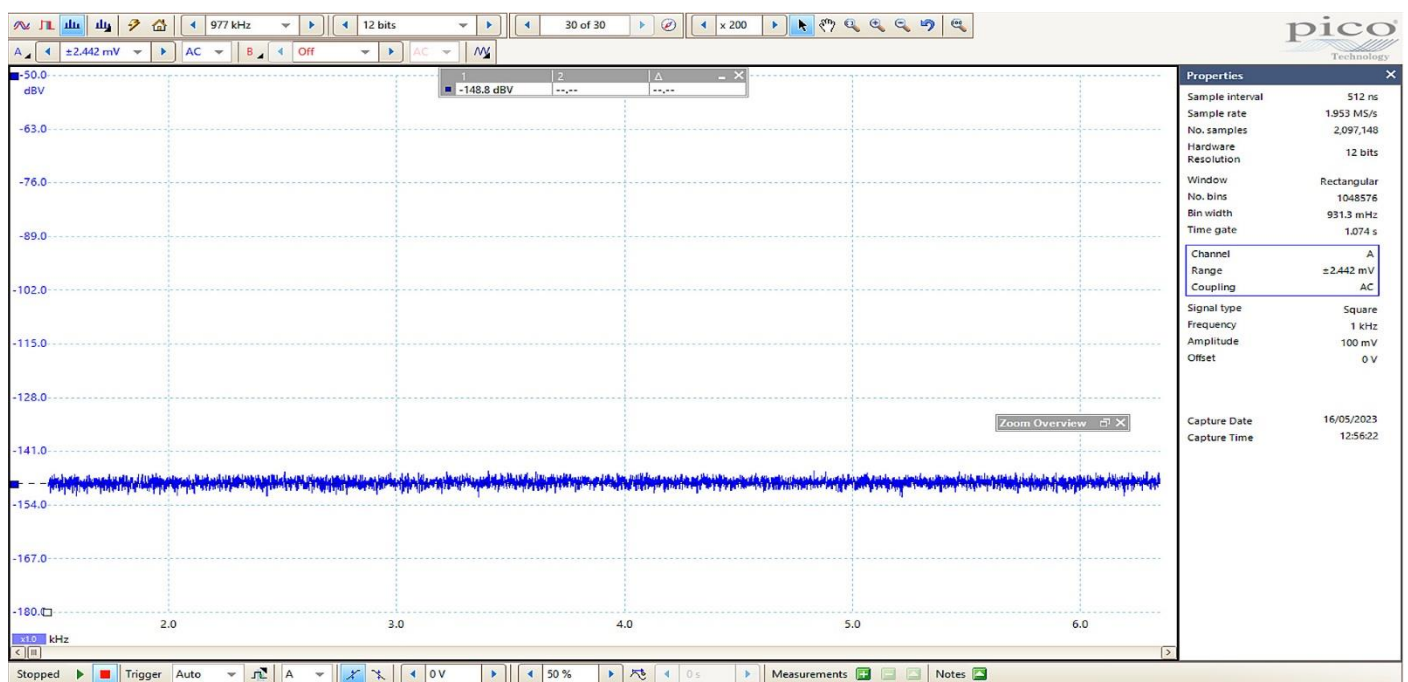


Fig 7, Noise level Silent track

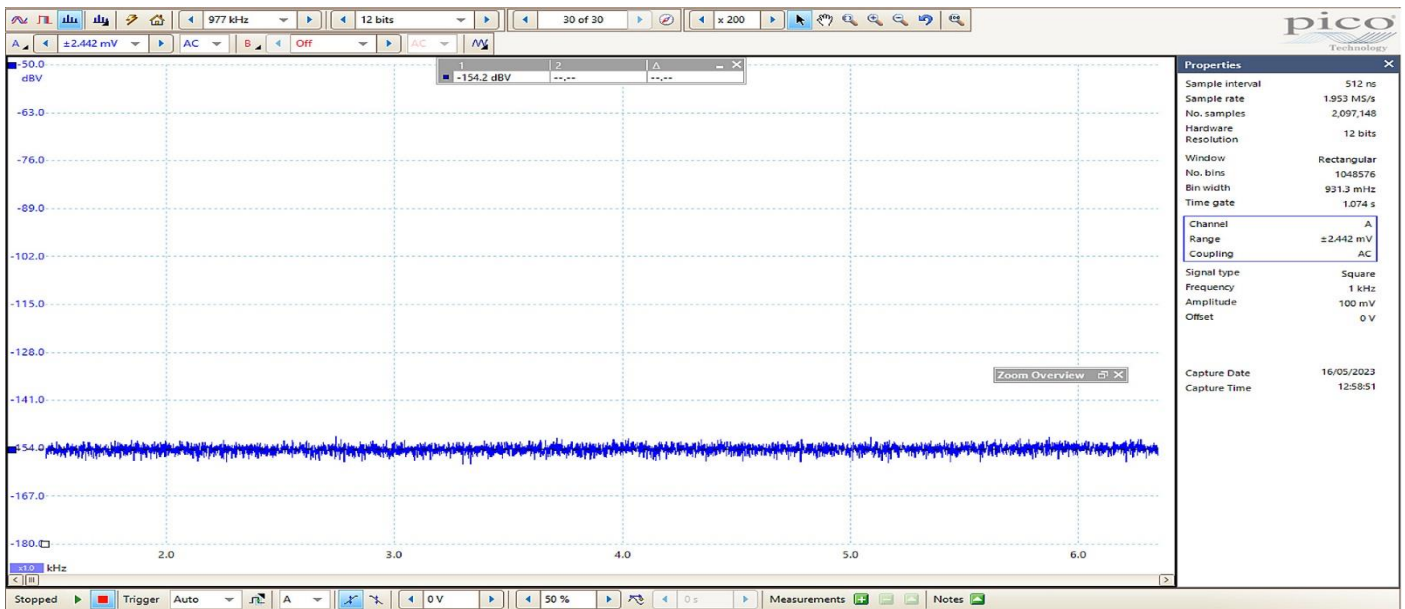


Fig 8, Noise level reconstruction filter, switch 1 open,

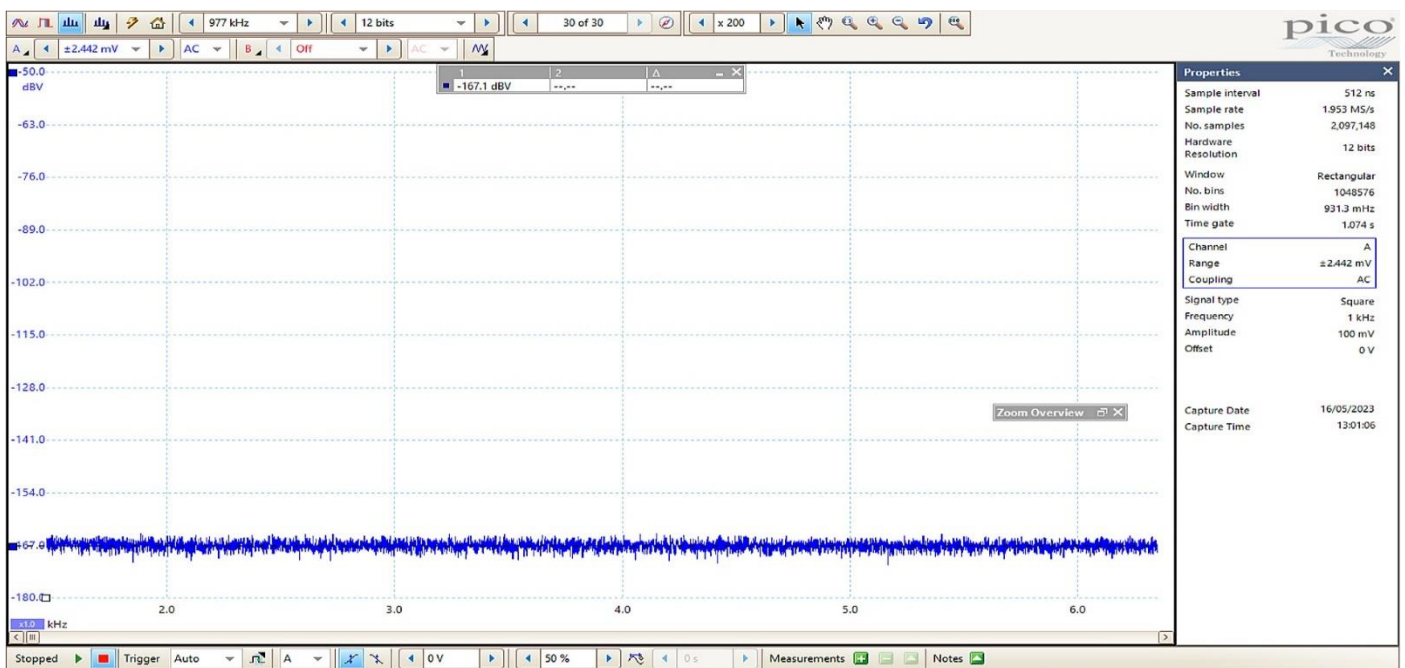


Fig 9, Noise level measuring system, terminated with same 2*50R at input as the Dac presents.

Calculating the Firdac's own noise in isolation, by taking the -154.2dB/rtHz reconstruction filter's noise from Fig 8 out of the measured overall -148.8dB/rtHz, gave 150.3dB/rtHz.

The reconstruction filter therefore is still worsening the Firdac's noise by 1.5dB from -150.3dB/rtHz to -148.8dB/rtHz.

Converting the -148.8dB/rtHz into S/N, means again adding $10 \cdot \log(20.000\text{Hz}) = +43.0\text{dB}$ to get a S/N of -105.8dB or 107.6dBA, a total improvement of 3.7dB over the original situation by just tweaking the reconstruction filter and bringing the load on the Firdac gates from 32k4 to 6K98.

Using an OPA2210 instead of the used OPA2134, an extra ca. 1dB could still be gained bringing Marcel's design to a S/N of -108.5 dBA thereby giving the digital part of the Firdac design all the credits it deserves.

Further results with the above mods.

Fig 10 shows the Dac's output with the same -90.31dB 24 bits@1KHz sine wave that Stereophile also uses in their tests.

Converting -90.31dBV into volts gives 30.5uV rms only 2% off from the average 29.88uV shown in the plot.

Because of above mentioned normalisation of putting 0dB at 0dBV, in fact the real rms value is $29.88\mu\text{V} * 1.967 = 58.77\mu\text{V}$.

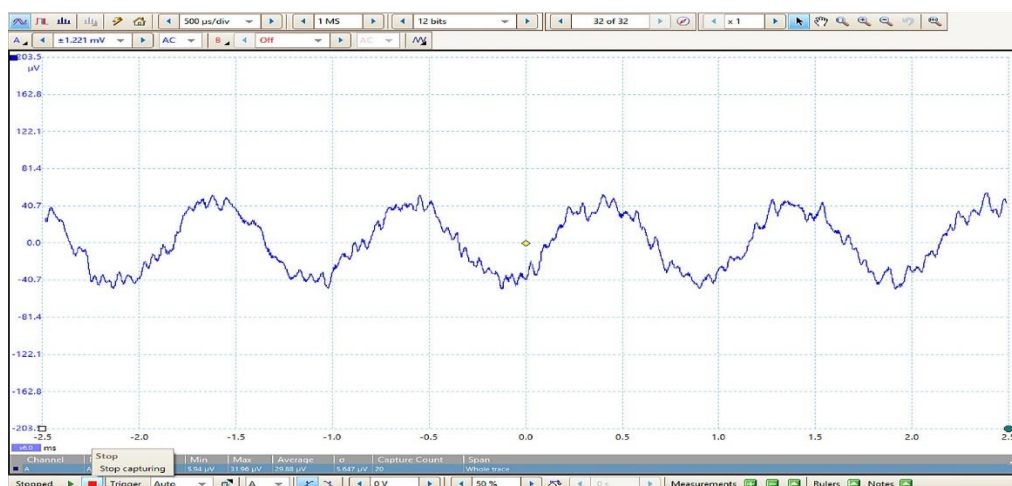


Fig 10, Normalised 24 bits 1KHz sine wave at -90.31dB

To complete measurements, THD and IMD tests were done as shown in Fig 11 and Fig 12.

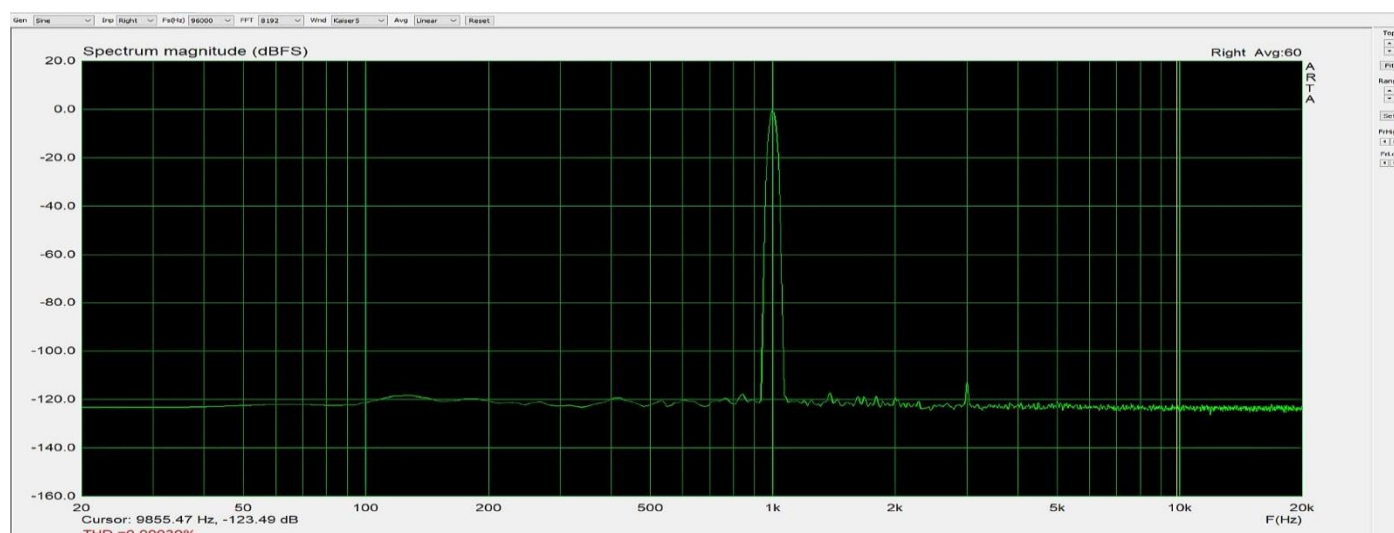


Fig 11, Incredibly low 3ppm distortion at 0dB@1KHz 16bits.

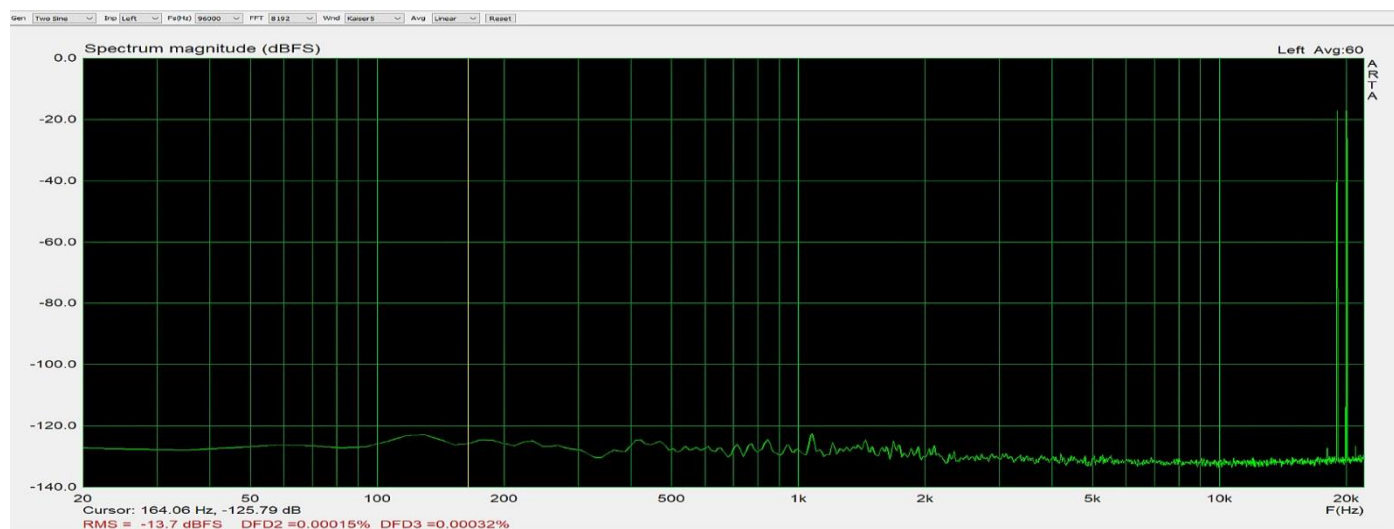


Fig 12, Very, very low IMD at -10dB peak

The lower noise modulation is the better, because it may well influence sound perception. In this case noise **decreases** by -1dB with raising signal level, which can be regarded as excellent. See measured results in table 1 below.

Input dB		Positive dB		Negative dB
0		-148,7		-148,9
-1		-148,3		-148,5
-2		-148,3		-148,5
-3		-148,1		-148,3
-4		-148,1		-148,3
-5		-148,1		-148,3
-6		-148,1		-148,3
Silence			-147,9	

Table 1, Noise Modulation measured with DC signals from +1Volt to -1Volt

Question however is whether all the exercises that I made, really turned into any auditory benefit. And that's exactly what the final listening test below will tell.

Sound Review

It's hard to judge any change in sound reproduction with the time between before and after the alterations, but my impression is that clarity without any sharpness and articulation had been slightly improved, but that may just as well come from the cap change on the LT3042's set pin.

The Dac however was still placing very much emphasis on the middle of the sound stage.

A singer, musically supported by a small group, was still placed more in front and the instruments farther backwards.

But the singer, male or female, sounded absolutely excellent.

Also my comment on the beginning of this review was still valid that classical music was still reproduced as if playing in a much smaller concert hall on a smaller podium.

So, a fantastic sounding Dac producing a sound stage that's perfect for Jazz and music from small groups, but for classical music it was missing the 3 dimensional impression of a great concert hall.

Final Mods

So all the above mods had not given any improvement on the restricted sound stage.

That's why as a last resort I decided to shorten the 2.2uF Audyn MKP output caps, also because the output offset voltage was below 2mV, and WOW, there it finally was, a full 3D sound stage that was missing so far.

I'm surprised what change caps in the audio path can make to the sound perception.

Just because the 2 caps and 1 resistor around the output buffer made no longer sense with the 2.2uF shorted, I made a direct connection between output and minus input.

This also caused an improvement in the audio's low end spectrum.

The final test was to compare "Blue in Green" from the Miles Davis on Columbia LP "Kind of Blue" against a CD made by Stereoplay in 2016 called "Vinyl Classics Vol 3.0" recorded directly from this very LP with their top of the bill reference gear. Comparing both takes the sound preferences out the equation and should show the Dac's transparentness.

I heard only very small differences, but my personal sound perception between both LP and CD was smaller than ever, again showing the quality of Marcel's Dac.

Resume

- 1) My current Dac is a Bel Canto 3.5VB equipped with the PCM1792, but in its final shape, Marcel's Dac improved on it by producing more clearness at the high end of the audio spectrum, without any sharpness. I can highly recommend this excellent product.
- 2) This Dac needs a very good low noise 15Volt supply, like with a LT3045 LDO regulator to prevent 50Hz AM modulation.
- 3) The LT3042, providing the critical 5V supply, needs an alu or tantalum electrolytic cap on its set pin, a ceramic cap is not suited for the job.
- 4) In its original version, the reconstruction filter is quite noisy, worsening the noise produced by the digital section by several dB's.
- 5) Not to change too much components, modifications were made that lowered the noise but must not be seen as the one and only solution, because the op-amps with this mod have to process signals that are 4 times as large. Because of that small IMD distortion emerged with 19Khz+20Khz signals both at -6dB. That's 10dB more signal as used in Fig 12 in the review.
Marcel's original noisy filter did not show any IMD at all levels.
However, the filter used for Marcel's RTZ Firdac has a much better noise figure because of A) the used OPA2210 for the first stage, having almost 4 times less voltage noise, and because B) of the much lower 3K01 Firdac resistor values instead of the 32K4 in this design. I already used 6K98 without any noticeable negative effects.
So, this shiftregister design seems the best option for the lowest additional noise and the lowest possible IMD.

- 6) Result of the reconstruction's filter noise reduction:

Filter	Noise	Noise	Noise	S/N
	Overall	Reconstr	Firdac	
	dB/rtHz	dB/rtHz	dB/rtHz	dBA
Original	-145,1	-145,9	-150,3	-103,9
12dB gain	-147,9	-151,5	-150,3	-106,7
6k98 resistors	-148,8	-154,2	-150,3	-107,6

Table 2, Effect on overall noise by reducing the reconstr. Filter's noise

- 7) Be very carefull when placing large caps in the Dac's output, because that may have a direct impact on preceived sound reproduction. Most likely you won't need them because of the very low output offset voltage of below 2mV. If so you can also simplify the output buffer by directly connecting output to minus input.
- 8) I have a preference for separated digital and analog ground, which is more or less the case with the Shiftregister RTZ Firdac. However given the quality of the reproduced sound, there was no reason to believe that with this design separating the two would have brought any benefit and the achieved S/N of around 108dBA seems more than adequate.