A prototype DAC to evaluate Crystal Semiconductor's CS4329

by

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Intro

In 1995 a friend of mine had obtained samples of the, at that time, newly released CS4329 DAC chip from Crystal Semiconductor Corporation. Soon after that I decided to build an evaluation board containing the CS4329 which I dubbed the Proto-(type)DAC. These early samples had a bug making the frequency response deviate +0.2 dB at 20 kHz but this is corrected in the current silicon. The object was to evaluate its potential to replace the trusty CS4328 I was currently using in the X-DAC 3.0, from Audio Crafters Guild, I had put together a year before. In short the CS4329 is a complete stereo DAC system featuring 20-bit resolution, digital interpolation filters, 128 times over sampled fourth-order delta-sigma modulators and switched-capacitor analog low-pass filters. To get the 2 Vrms of output, as is required by the CD-standard, from a single 5V-supply the CS4329 incorporates differential outputs, i.e. two signals of 2.8 Vpp per stereo channel having a 180 degree phases difference and riding on +2.2 V of DC. Summing these signals in a differential amplifier yields 5.6 Vpp (which equals 2 Vrms) and the DC-components cancel as they are of equal magnitude and polarity.

Features of the Proto-DAC

The Proto-DAC is housed in a black coated all-aluminum cabinet measuring 260x380x90mm (wxdxh). It is built from four U-shaped profiles which mitered joints are held together with L-profiles in each corner, much like the frame surrounding a painting. Take a look at Fig. 1 as I describe the features of the Proto-DAC. Starting by way of its printed circuit boards, the digital mother board is the main pcb. It hosts a BNC input jack and two RCA output jacks, two input transformers, the Crystal CS8412 receiver (the E-revision) and the CS4329 DAC, analog filters and various voltage regulators. The digital mother board and the two analog filter boards are double sided with ground planes. The power supply board, display board and the eight plug-in regulator boards are single sided, I will return to these later on. From the BNC-jack the S/PDIF-signal is routed through either a high quality Scientific Conversion SC944-05 1:2 transformer or a cheap plastic cased Pulse Engineering PE65612 1:1 transformer. The transformers are selected with two relays by means of a switch on the display board mounted behind the front panel. The transformers pass the S/PDIF-signal to the CS8412 which has separate regulators for the digital section and the PLL (phase locked loop). These regulators are mounted on small pcbs with pin connectors that plugs into the digital mother board and the analog filter boards. The regulators are Linear Technologies LT317/337 AT at the moment, but they may be exchanged easily.

Figure 1 shows the interior of the Proto-DAC, sans the plug an play board containing the AD1890/91 sample rate converter, oscillators and dividers.

The plug and play feature

The four signals (serial data, master clock, serial clock and left/right-clock) are passed from the CS8412 to the CS4329 by means of four relays, either directly or via a 40-pol DIL-connector which may be used to plug in various devices between the CS8412 and the CS4329, such as:

- Flip-flops to re-clock serial data, serial clock and left/right-clock with the master clock;

- Sample rate converters such as AD 1890/91 together with oscillators and dividers;

- A FIFO-buffer and a VCXO together with associated logic.

The 40-pole DIL-connector also provides two independent 5V supplies and a reset-signal to the AD1890/91. The plug and play device between the CS8412 and CS4329 may be switched in or out with a switch on the display board at the front panel. The CS4329 may run either with a serial clock signal from the CS8412 or it may generate it by itself (this is described in more detail later on). This is accomplished with two relays: one connects the serial clock-pin of the CS4329 to ground and the other relay is used to momentarily disconnect the left/right-clock signal from the CS8412. Two switches on the display board provides for this feature. The CS4329 has separate regulators for the analog section and digital section. The ground plane is common to both the analog and digital section.

Taking care of the analog output

The differential outputs of the CS4329 are routed to two analog filter boards, one for the left channel and one for the right channel, which plugs into the digital mother board. Here the positive and negative analog outputs are summed and low-pass filtered. The circuit is mostly the same as the one published in the "Crystal design note for a 2-pole filter with differential input" by Steven Green, se Fig. 2. The op amp is an Analog Devices AD797 BN which output is buffered by an AD811 AN, working as a follower. By means of a relay the feedback to the filter may either be taken from the output of the AD797 or the AD811 which enables a choice between local or global feedback. A disadvantage of the differential circuit may be the loading effect it provides on the DAC-outputs (about 14 kohm from the filter). This low value may impair CMRR as there is little control over the source resistance's of the DAC-outputs. A small mismatch in these resistance's may degrade the CMRR of the filter circuit. A cure for this is a very high impedance at both filter inputs. This is accomplished by a circuitry lifted from Walt Jung's IC Op-Amp Cookbook, JFET-followers with matched dual-JFET cascode pairs and a high-CMRR current source. This buffer may be switched in or out with two relays. Each analog filter board has plug-in regulators supplying +/- 15V.

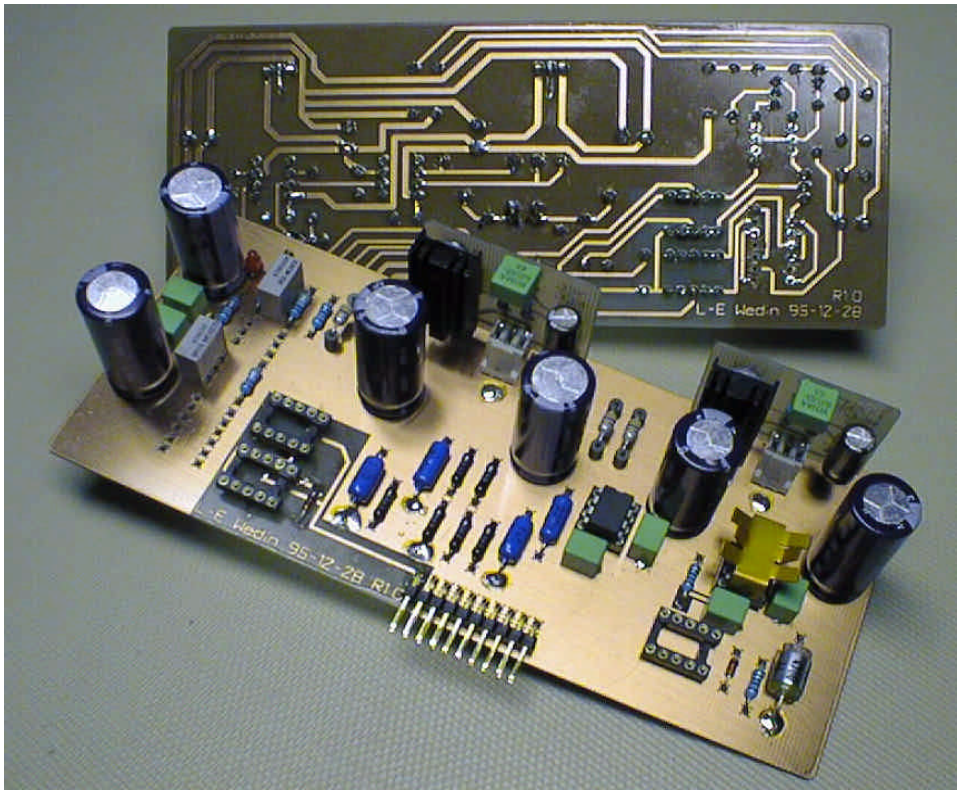

Figure 2 shows the prototype implementation of the 2-pole low-pass filter with differential input as is recommended by Crystal Semiconductors, notice the heat sink on AD811. Relays and input buffers have been removed from this pcb.

Power supply and front panel options

The power supply board has a simple common mode mains filter and three pcb mount potted UI-core transformers, each having dual secondaries. One transformer is dedicated to the analog filters, the other to the CS8412 and CS4329. The third may be used by the plug and play board between the receiver and the DAC. Each secondary has a bridge rectifier with four GI856's and a pi-filter with electrolytics and a small 1mH choke. There is also a 24V supply for the relays, which counts to a total of 14.

At the back of the front panel is a display board with three LEDs and seven switches. The LEDs indicate power on, de-emphasis activated and the absence of a signal from the transport or that the DAC has been muted. The switches have the following functions:

- Global or local feedback in analog filters;

- Buffered or unbuffered DAC-outputs to analog filters;

- Internal or external generation of serial clock;

- Disconnecting left/right-clock (to be used when switching from external to internal serial clock);

- Asynchronous or synchronous DAC-interface (switching the plug and play board, between CS8412 and CS4329, in or out in case it carries an AD1890/91);

- Switching between S/PDIF-transformers;

- Mute/calibrate/reset of the CS8412, CS4329 and AD1890/91 if used.

The all important analog filtering

Well, what have I learnt from listening to the Proto-DAC? This: the thing that matters the most, sound wise, is by far the analog filtering of the CS4329! The CS4328 has an analog filter consisting of a 5th order switched capacitor low-pass filter and a 2nd order continuous time filter. The switched capacitor filter eliminates out-of-band energy resulting from the noise shaping process and the continuous time filter eliminates high frequency energy that appears at multiples of the 64x over sampling. The clever thing with the CS4328 is that its 64xfs (where fs is the audio sample frequency, i.e. 44.1 kHz for CD) digital interpolation filter equalizes the phase response of the analog filter producing a resulting phase response deviating only 0.5 degrees from linear phase. The CS4329 has an 128xfs digital interpolation filter and an analog switched capacitor filter but does not include phase or amplitude compensation for an external analog low-pass filter. Hence, the DAC system's phase and amplitude response will depend on the external analog conditioning.

Analog conditioning

The sound produced from the 2-pole filter with differential input from the Crystal design note was disappointing, to say the least. Removing the filter capacitors and running it as a bare summing amp was a great revelation, but I was not quite content with the sound of the op amps and I also thought that the digital noise should be suppressed. (It sounds great, but looks terrible when viewed on an oscilloscope.) So I designed a single ended class A summing and filtering amp with discrete components which I think sounds just marvelous. In short, it features a dual JFET as the differential amp and a MOSFET as buffer, both are driven by current sources. It has proved to be quite sensitive to the choice of passive components and it has been very educational to listen to substitutions of various resistors and capacitors. As you may see from Fig. 3 I am currently using Vishay resistors and MIT Multi Cap and Rel-Cap Teflon capacitors at the output. The electrolytics are Sanyo's OS-CON series which is an aluminum solid capacitor with an organic semiconductive electrolyte (a complex salt in the form of black fine-powdered crystals). The greatest advantages of the OS-CON is its low resistivity and good high frequency characteristics, nearly equal to that of film capacitors. The audio grade SG-series are resin sealed to reduce microphonics or so called element vibration and have leads of oxygen free copper.

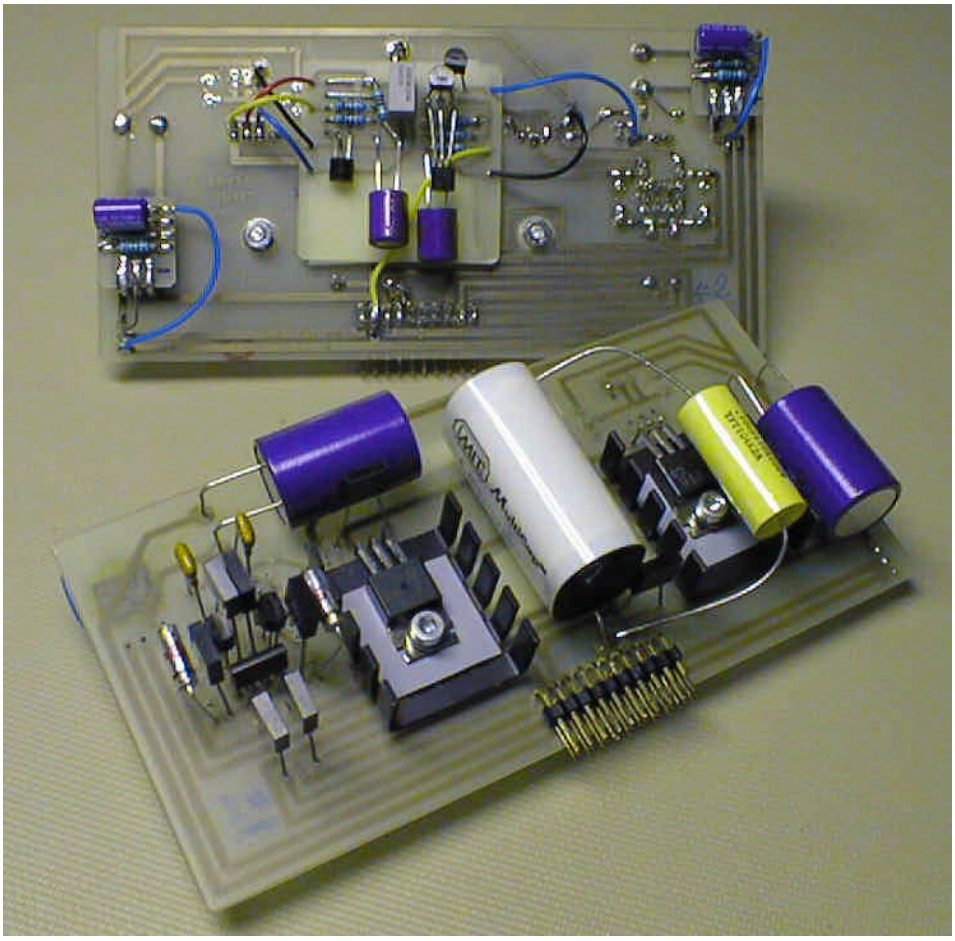

Figure 3 shows a prototype implementation of the single ended class A summing amp I came to like very much.

Power supply rearrangements

The original Proto-DAC had independent regulators for the analog and digital supply on both the CS8412 and CS4328. However documentation presented at the Crystal World Tour Application Seminar in 1996 revealed that there existed recommended operating conditions for Crystal Semiconductor's ICs which specified power supply voltages. These specifications dictated that the analog supply must always be the most positive voltage on the part. When the digital supply is separately regulated, care should be taken to ensure that the analog supply always comes up first and never drops below the digital supply. This fact made me rewire the supplies in a way to ensure that the analog supply is always greater than digital supply, i.e. I derived the digital supply from the analog supply through a small inductor and a capacitor. The inductor and capacitor combination provides a low pass filter to isolate the digital switching noise from the analog power supply. The decoupling capacitors should be located as close to the CS8412 and CS4329 supply pins as possible.

Did I hear any differences?

Another improvement to the Proto-DAC was switching to a newer G-revision of the CS8412. From that revision on the silicon is made in a different manufacturing process, making it smaller and also improving jitter specifications. I also placed a 1.0 nF capacitor between the PLL filter pin and ground to improve jitter performance even more. Well, what about hearing any differences? To be honest, apart from the analog filters I have heard no audible difference. Switching between the two S/PDIF-interface transformers revealed no differences in sound reproduction. I have also

heard no audible difference when running the CS4329 in external or internal serial clock mode. In the internal serial clock mode, the serial clock is internally derived and synchronous with the master clock. The use of the internal serial clock mode simplifies routing of the pcb layout by allowing the serial clock trace to be deleted and avoids possible interference effects which may degrade system performance. Crystal also states in their application recommendations that the CS4329 should be operated in internal serial clock mode to obtain optimal performance. Unfortunately I never got to compare synchronous and asynchronous operation of the Proto-DAC. I built two versions of a plug and play board containing an AD1890, ECL-oscillator and MECL-dividers to obtain the left/right-clock and bit clock from the oscillator but never got it to work all right (pops and clicks most of the time). I understand MECL-chips are very sensitive to pcb layout and proper termination. I pondered a third version with ECL-oscillator, an LT1016 comparator to convert the ECL-levels to TTL logic levels and a 4040 divider to produce the left/right-clock and bit clock but somehow never got the time to implement it. No doubt, the Proto-DAC's not quite optimal pcb layout has contributed to make it difficult to "hear the differences". This is in part a consequence of the number of features implemented in the Proto-DAC and the fact that signals with fast rise and fall times have to pass through several relay contacts.

Outro

However, in retrospect I think the Proto-DAC has served its purpose well. So, what are my plans for the future as the Proto-DAC was just an interim solution? Well, I'm going to design an analog board containing the single ended class A summing amp and the CS4329, or maybe the 24-bit CS4390 which is pin compatible with the CS4329. The S/PDIF-input receiver and stable clocking of the serial data signal, left/right-clock and master clock signals will be handled by a separate receiver board. The two boards will be interconnected by thin coaxial cables and miniature pcb mount BNC-connectors. This way, different receiver boards and analog boards may be exchanged quite easily (as long as they have the same interfaces and physical size). With this flexibility together with a well filtered shunt regulated raw supply and a solid aluminum case, I think I will be rewarded by many hours of happy listening in the years to come.