Jitter in (Digital) Audio

1. Summary

Jitter is not different from phase noise. Jitter is defined in the time domain and phase noise in the frequency domain. They can't be related to each other directly because jitter does not define the frequency at which the (digital) signal jitters.
It is known rather shortly that the eventually audio quality can be influenced heavily by jitter on the digital audio signal and its synchronizing clock. In this paper we will see when and how jitter contributes audio quality. It turns out that close in phase noise in the digital domain is the malefactor. Phase noise at larger distances from the carrier will worsen the signal to noise ratio (S/N) but does not introduce the kind of distortion for which the human perception seems to be very sensitive.

Another phenomenon of jitter, which is hardly known, takes place in the analogue domain. I mean the influence of (large) low frequency signals (typically 100 Hz) on higher frequencies (say 5 kHz). Here non linear junction capacities are mainly the malefactor.

The kinds of distortion caused by the two jitter-phenomena sounds fairly similar. Even stronger: (digital) clock-jitter causes audio jitter, so that it sometimes will be difficult to find the reason. Often both of them are present.

2. Jitter in the digital domain

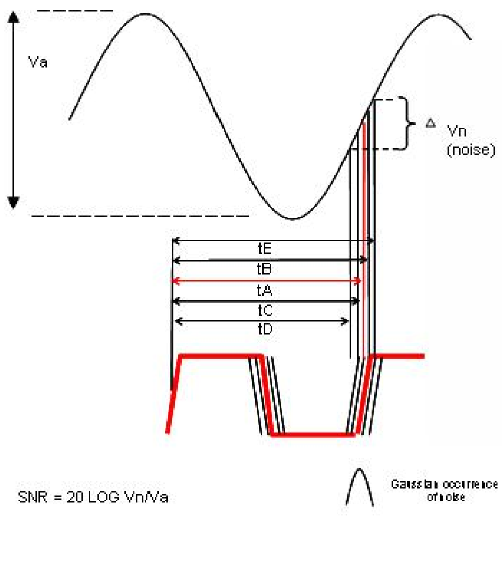
Digital audio data signals are defined by its word length and bit-frequency. For high-end audio the word length is at least 16 bit with a sampling rate of 44.1 kHz. The word length restricts the dynamic range (16 bits corresponds with 96 dB) and the sampling rate restricts the bandwidth (44.1 kHz corresponds with 22 kHz [Niquist]). Nowadays often 24 bits and 48 or even 192 kHz are used.
The signal does have a format in which it is stored. MP3 is a well known format for medium quality music. WAV is a common PCM-format for high quality. There is a great number of different formats (like SSD) that will be left aside here.
The serial interface: SPDIF (or S/PDIF) is the Sony/Philips Digital Interconnect Format which serves connection between audio equipment.
I^2S, IIS, Inter IC Sound or Integrated InterChip Sound is a serial bus interface standard used for connecting digital audio devices, generally within one cabinet.
All these signals will suffer from jitter. If, however, the jitter is much smaller than the bit length, the integrity of the (audio) data will not be corrupted until it is transformed (back) to the corresponding audio signal. In case the signals have to be synchronised by a so called clock signal. Also this clock signal will suffer from jitter.

When does jitter affect audio quality?

Every audio chain starts, sooner or later, with a (small) electrical analogue audio signal from a microphone and ends with a (large) electrical analogue audio signal into a loudspeaker system. With a public address installation this event is real time: the microphone transforms the audio into an analogue electrical signal which will be filtered, and compressed (if required) and amplified to the electrical power needed for the speakers which will transform the electrical signal into (stronger) audio again.

With reproduction systems, the audio has to be recorded. Until the 80s of the last century this recording has been an **analogue** affair. The storage medium was analogue: tape or a gramophone record.

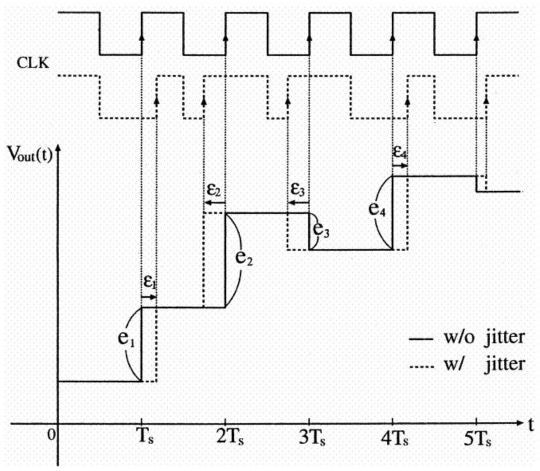
Nowadays the storage of audio happens on digital storage as CD, digital tape or disk or whatever digital storage medium, so the audio has to be converted from an analogue signal into a digital signal by a so called **ADC**: an Analogue to Digital Converter.
For reproduction the digital recording should be 'played'. In this player the digital signal has to be converted (back) into the corresponding analogue audio signal by a **DAC**: Digital to Analogue Converter.



As long as the audio is treated in the digital domain, jitter will not be so important, **but** when conversion from analogue to digital and digital to analogue takes place, jitter becomes extremely important!

The ADC

An ADC converts an analogue electrical signal of some volts into a digital signal. For this the audio signal has to be **sampled**. How precise the obtained digital signal will represent the original analogue audio depends on the sampling rate and **the jitter** on the sampling signal. This sampling signal is supplied by a so called **clock oscillator**. A noisy/jittery clock signal will distort the digital audio representation, which errors can't be corrected later on!



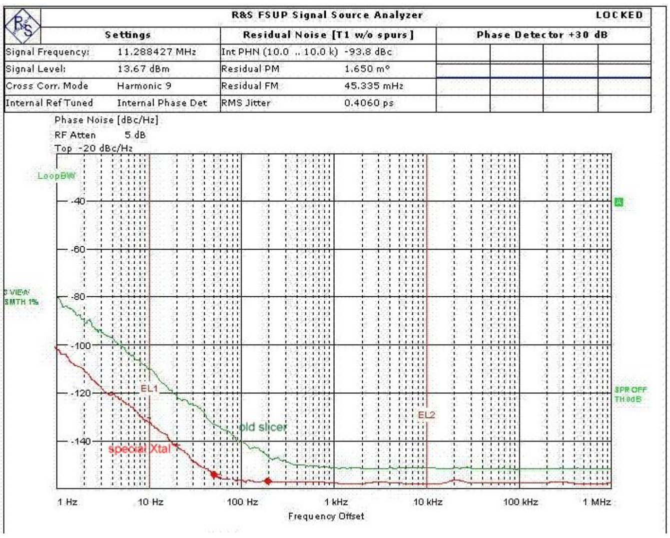
DAC output waveforms with ideal sampling clock (without jitter) and actual sampling clock (with jitter ε_n).

The DAC

The DAC converts a digital signal into an analogue signal (often an electrical current of a few mA). For this the digital audio signal should be as clean as possible. Any jitter/noise will distort the analogue representation of the digital signal. After transporting the digital signal from one box to another and/or make all kind of calculations and transforms, (digital alias filtering,) the signal will show up too much jitter. This will be demonstrated into a very unlikely distortion in the higher audio region which also can't be corrected later on any way. This kind of distortion is often called 'digital audio distortion' in the audio scene.

The digital data signal however could be 'reclocked' with a jitter-poor clock oscillator in a 'reclocking-circuit' before it enters the DAC-chip!

N.B.: HF-jitter on the clock signal distorts the eventually audio signal in such a way that the zero-crossings are dislocated. If the HF-jitter on the clock signal is audio related, the kind of distortion is similar to jitter in the analogue domain (see later) for which human perception is very sensitive! So, one could imagine that close in phase noise is the most miserable.



The Clock Oscillator

With AD- as well as with DA-conversion a clock oscillator 'of good quality' should be used. But what is good quality? Well, the clock jitter should be < 0.5 ps. But this is not all. Recently phase noise close to the carrier ('close in noise') below about 100 Hz turned out to be the most miserable! It blurs the stereo image: the voices and instruments become wider and the sound stage becomes less deep. The phase noise of the clock should not exceed -130 dBc@10Hz! In the picture left the green curve belongs to the header of the R&S analyser. That oscillator had been equipped with an LT1016 as a slicer. The red curve has been measured with a DC-receiver. The Xtal had been selected and the slicer (= sine to logic translator) was a 74HC04. This clock oscillator is that good that it is implemented into the CC1 of Grimm Audio! (http://www.grimmaudio.com/pro-products/master-clocks/cc1/)

Noise at, say, > 1 kHz from the carrier will of course degrade the signal to noise ratio but will not lead to the typical unfavourable harsh sound and/or bad stereo imaging.

Some people think that the *absolute frequency* of a clock oscillator is important. The clock frequency determines the pitch of the audio. A deviation of 4 Hz at 1000 Hz however can hardly be detected by humans so that the absolute frequency could deviate 0.4% which is 4,000 ppm….! In a commercial environment 100 ppm has been prescribed.

3. Jitter in the analogue domain

A total different phenomenon is jitter of higher audio frequencies caused through (stronger) low frequencies in the same audio signal. This is different from intermodulation and phase modulation (in my opinion). Humans are very sensitive for this kind of distortion. Do not ask me why! This was discovered by Henk ten Pierick †. Moreover he developed a method to rank circuits: tubes, discrete semiconductor circuits and op amps, on audio quality by measuring this jitter.

This kind of distortion is caused by voltage dependant delay in (analogue) circuits, which mainly happens because of non linear parasitic capacitances in junctions. In general: use as small (HF) semiconductors as possible, use high voltages and low impedances.

Op amps like the LT1028 (for virtual ground) and OPA134 score high in quality at this point. In exceptional cases *bootstrapping* could be a solution. In a number of articles on this website these items are discussed.

4. Remedies

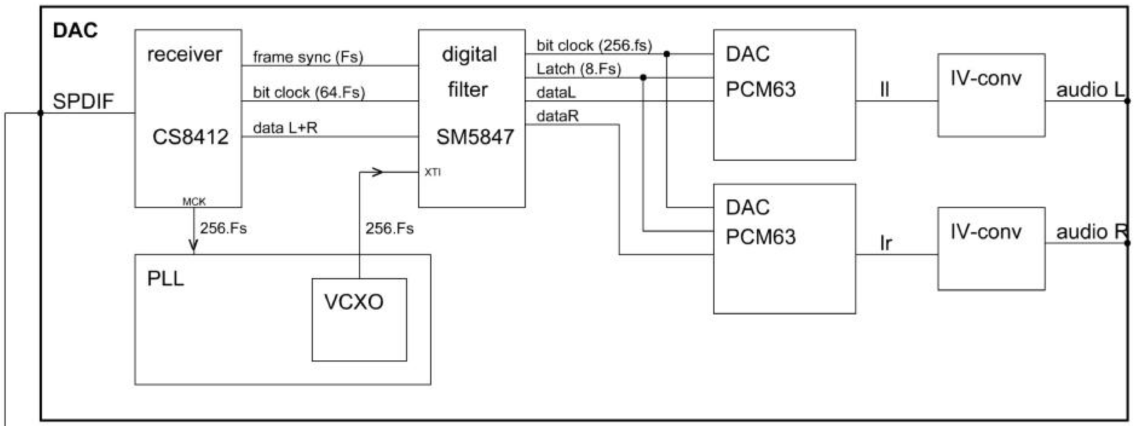
For a **good clock oscillator** (XO) one should dig into the articles: "Reproducible Low noise oscillators" and “The best clock oscillator” on this website.
For **bootstrapping**: "Condenser microphone pre-amp with bootstrapped op amp."

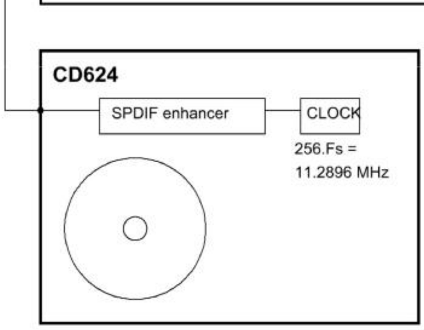
'Re-clocking'

For re-clocking the digital audio signal, the bitclock and the LATCH should be re-clocked before it enters the DAC-chip. One should place the clock oscillator as close to the DAC-chip as possible in the same box. Moreover the synchronizing signals should be kept separated from the data to avoid jitter cross talk (audio-related jitter is the most bad jitter! [Henk ten Pierick]). The averages of eg. the SPDIF signal contains low audio frequency information (listen to it with a headphone)!
If a separate DAC is used in combination with a CD 'transport', the connection between the both is not standard any more! The SPDIF signal goes from transport to the DAC and the clock-signal from DAC to transport as shown in the next circuit diagrams.

5. Examples

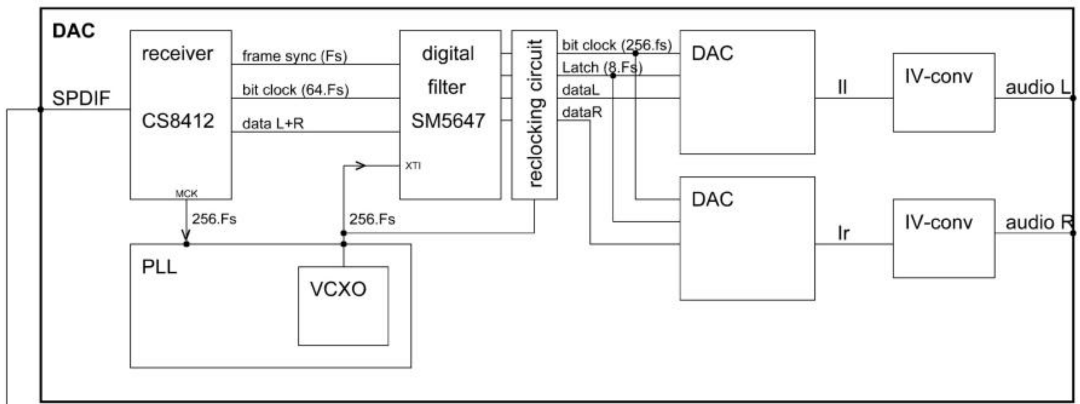
In the figures below the connection of a CD-transport to a DAC has been expounded. Each next step offers better jitter conditions.

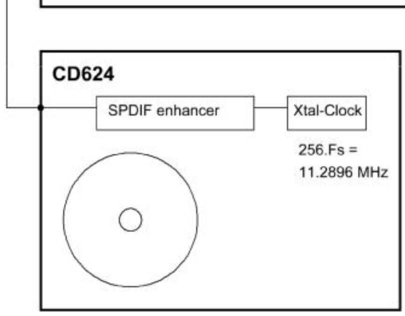




Before the noisy clock signal from the receiver enters the digital filter, it is enhanced by a PLL with a VCXO.

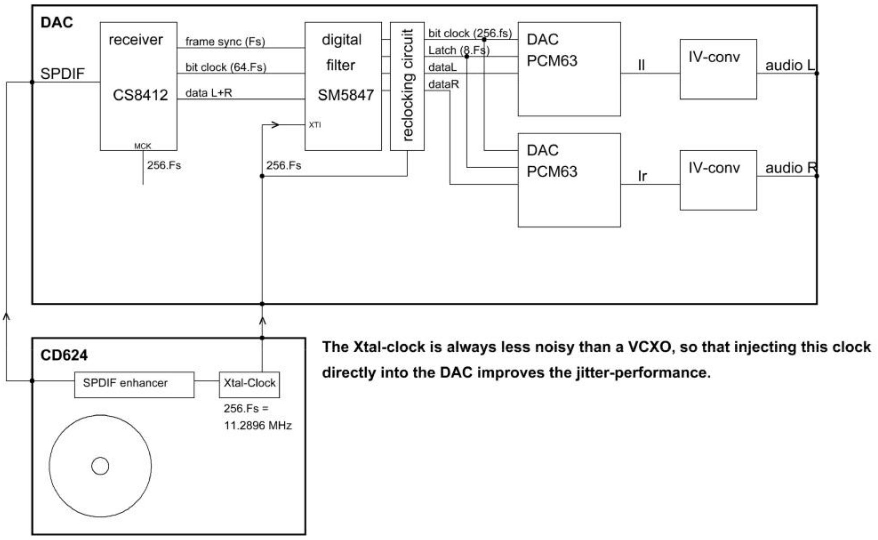
The noisy SPDIF signal in the CD624 transport is enhanced as well.

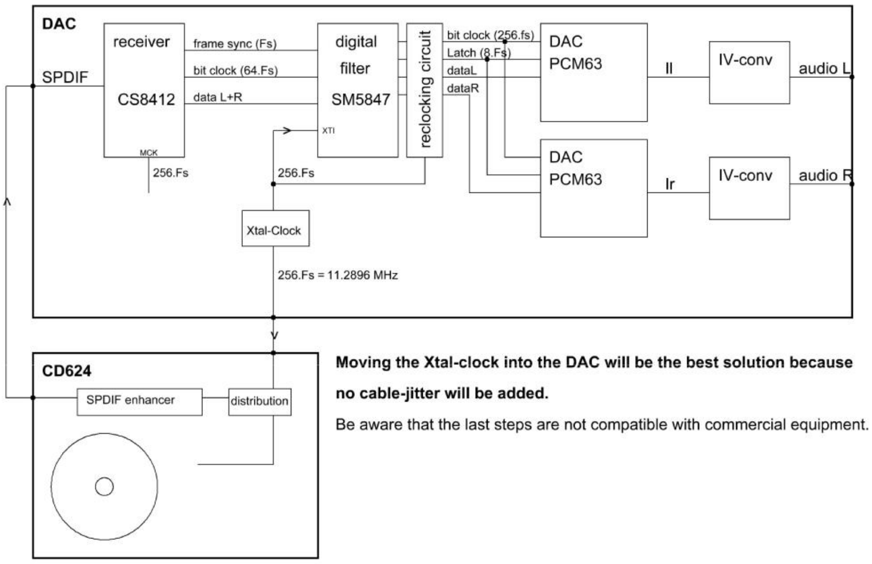




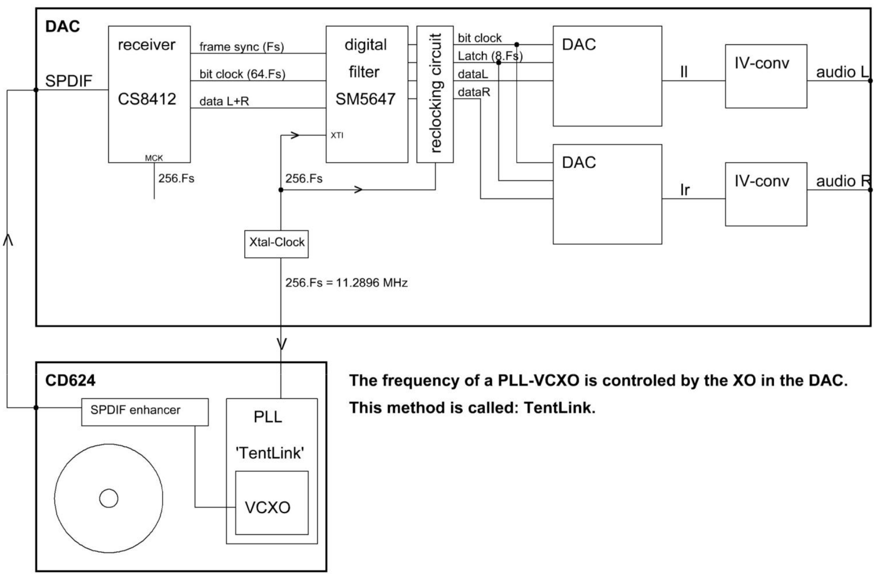
The noisy signals from the digital filter are reclocked by the 'reclocking circuit', directly controlled by the VCXO, before they enter the DAC's.

The clock- and data-signals should be re-clocked in different circuits, so that no cross talk will take place.



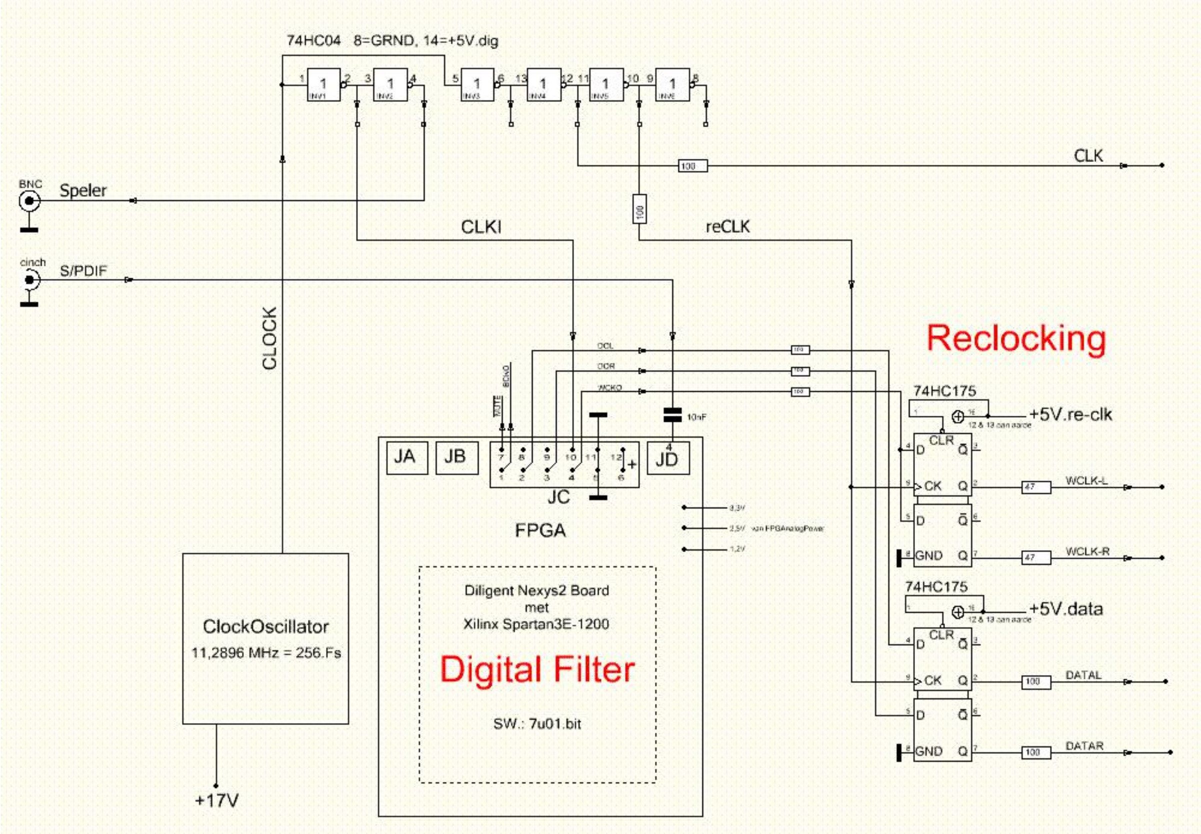


With this last solution the connection of the Xtal-clock from the DAC to the transport must be secured, because most transports will malfunction if powered on without a clock signal. A good solution could be to accommodate the transport with a VCXO which is synchronised by the clock signal from the DAC as with the so called ‘TentLink’. The circuit is available with TenLabs.
In my case, I powered the CD624 from the DAC, so it switches on with the same mains switch.

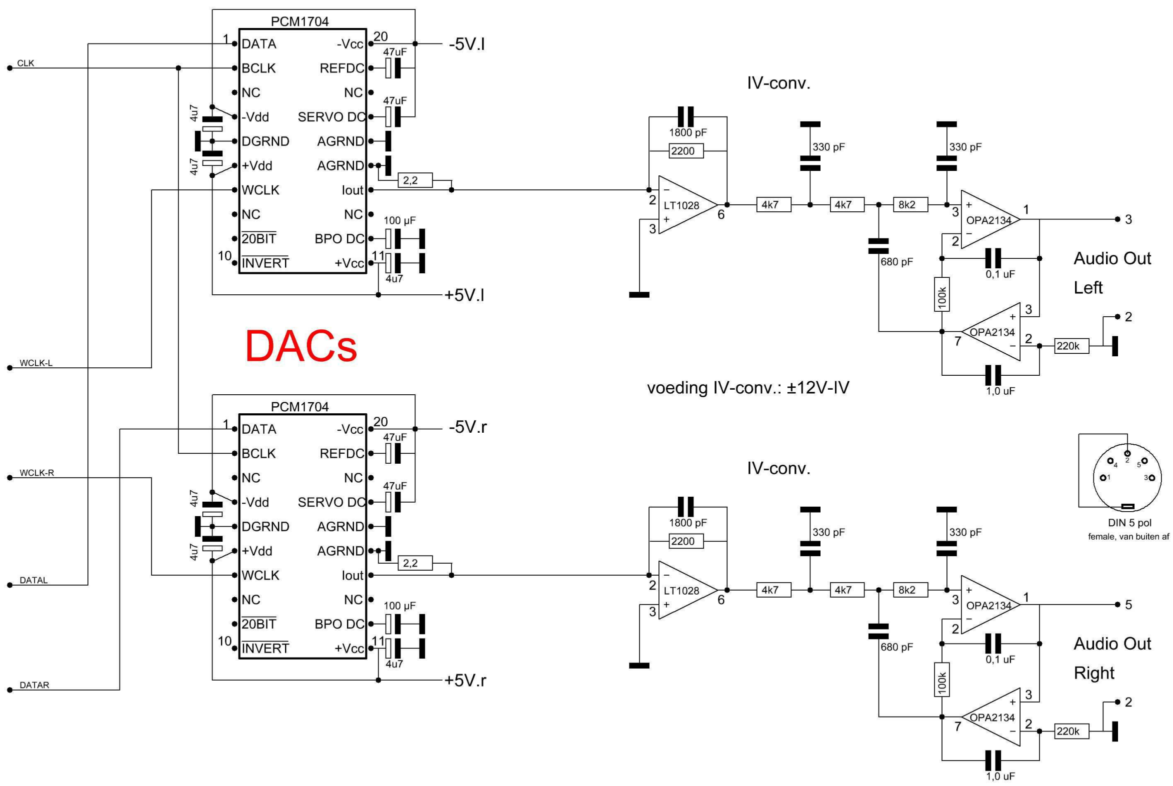


6. Some of the measures

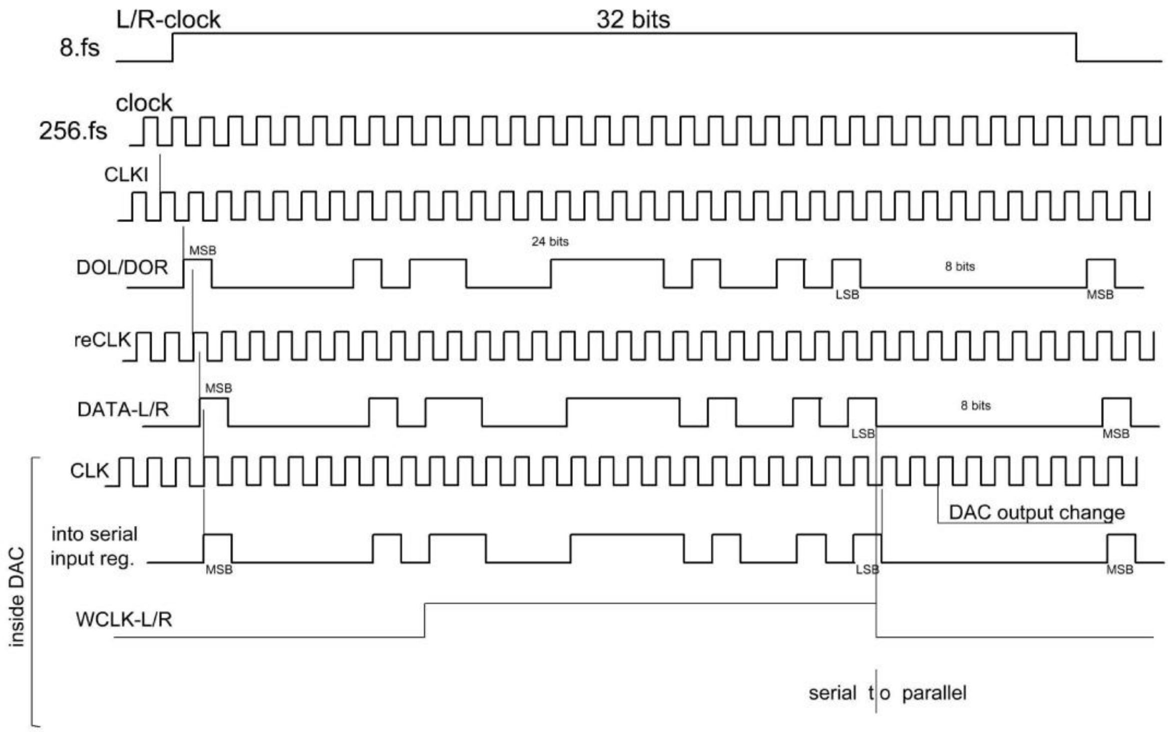
Of the penultimate proposal, some detailed circuit descriptions are shown (with different components). Mind that the DC-supplies to nearly every circuit are separated: +5V.dig, +5V.re-clk, +5V.data, etc.



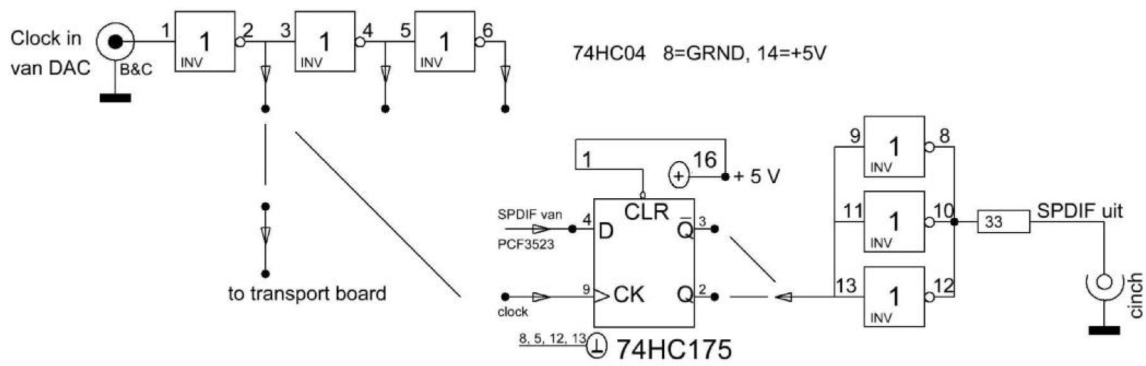
With ‘Speler’ is meant: Transport. The Digital Filter in case is an FPGA with the diy software 7u01.bit. The upper 74HC04 serves **timing!**



Timing of the different signals:



SPDIF enhancer in transport



For circuits to avoid **'jitter in the analogue domain'**. Look at the articles:

"Another 35 watt Solid State Amplifier", and

"Condenser microphone pre-amp with bootstrapped op amp".

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P.S.: With the DAC: **PCM1792** this reclocking is not needed because it has been implemented into the chip itself! The only precondition is that the bit clock (BCK, pin 6) has as little jitter as possible so put the Xtal oscillator on 11.2896 MHz close to the DAC, at least in the same cabinet.