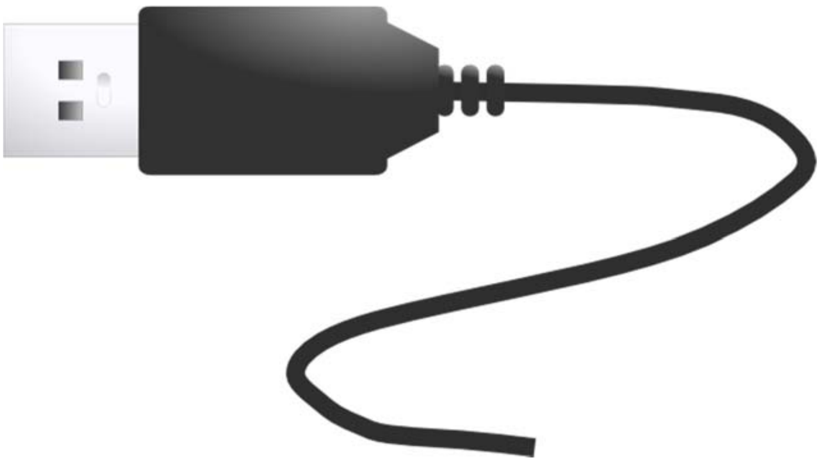
ZeroUno DAC

USB INTERCONNECTION



INDEX

INTRODUCTION

This tutorial is dedicated to the users want to discover the ways to set up the audio playback software for the best interface with ***ZeroUno*** **DAC**.

It is intended for newbies and many things are written to explain to the users without any skill about computers setup and/or the audio playback software setup.

Sometimes the terms used should be not perfect from the academic point of view, or sometimes the explanations will be too superficial.
This because the tutorial is addressed everybody.

At the base of all the audio formats there is the Nyquist-Shannon sampling theorem.
Harry Nyquist and Clause E. Shannon demonstrated that to do not lose any information of an analog signal with a define band, the sampling frequency must be double the width of the band.

The theorem and the human hearing capability of 20KHz and 80dB of dynamic range let become very popular the sampling frequency of 44.1KHz, just a little over 2 times 20KHz, and a dynamic range of 96dB possible with 16 bit (only) of resolution.

Starting from this Sony and Philips in the 1982 defined the format CD-DA for the Compact Disk, and with a sample rate of 44.1KHz is possible to sample (analog-to-digital and digital-to-analog) band of frequencies up to 22.05KHz.

These 22.05KHz is the maximum frequency band with 44.1KHz of sampling rate.
This band should be compared with the conventional upper limit of human hearing that is 20KHz at one month old.
Anyway, for the purists, in the 2014 a research showed that human brainwaves show responses to hearing frequencies at 32 kHz.

16bits of resolution for each sample get a dynamic range of 96dB that should be compared with:
- the dynamic range of human hearing that is roughly 140 dB
- the dynamic range of music as normally perceived in a concert hall doesn't exceed 80 dB
- the human speech is normally perceived over a range of about 40 dB.

Usually the low quality of some CD is not caused by the limit of 16bit - 44.1KHz but by the poor job done at the recording studio when the format was generated.

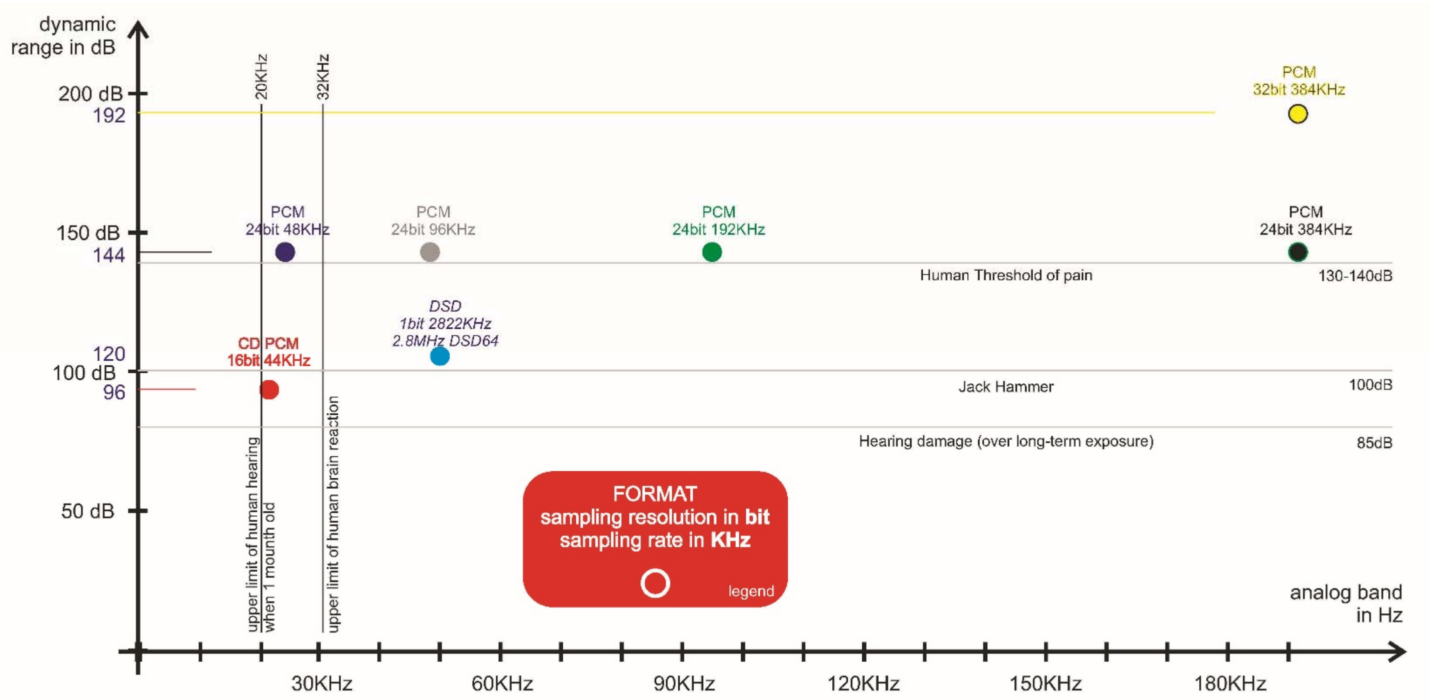
Starting from the 1982 a lot of other audio formats appeared on the market, with the target to exhibit better performances.

- 1995 - HDCD with 20 bits resolution and a theoretical dynamic range of 120dB.
- 1998 - DAD-Audio with max 24 bit – 96KHz then become 24 bit – 192KHz.
- 1999 - SACD based on the DSD sampling method opposite the PCM(*), able of 50KHz audio band (about 100KHz unfiltered but not usable) and a dynamic range of 120 dB in the 20-20KHz audio band (**).
- 2000 - DVD-Audio with max 24 bit – 96KHz then become 24 bit – 192KHz.

…. none of above audio formats achieved widespread adoption.

In the 2008 HDtracks introduced the download of high-resolution audio generating a total revolution cause the no need of a physical support for the audio albums and its player cancelled, on the other hands HDtracks is not an electronic manufacturer.

From the CD starting point is useful to use a graph to compare the most popular audio formats. The reader can chose the better for its needs.



The right format to reproduce an album is out of the scope of this document, in terms of either bits and / or sampling rate and / or PCM DSD.

The quality usually does not depend on the audio format implemented but on the job done by the recording studio.

(*) A pure DSD tracks but the raw master format probably does not exsist:
in a recording studio the signals in the raw master are mixed, faded, equalized, reverb added, an so on, so all of the DSD signals are transcoded into PCM (or analog) signals, then the signal processing applied, and finally re-modulating the signal back into DSD.
(**) Define the resolution and the band of the DSD is not possible like for the PCM.
DSD works in a totally different way and the sampling process generate a big content of unwanted frequencies over the useful band, so a filtering/decimating process is needed.
It is not a defect, it is a characteristic of the process. So a DSD64 process generate a raw band of about 100KHz but a lot of noise in introduced in the band over 50KHz and it must be filtered. Therefore, the useful band is about 50KHz.

USB INTERCONNECTION – PCM, DSD, DoP FORMATS

The *ZeroUno* **DAC** is equipped with an USB 2.0 input able of *USB Audio 2.0* for a high performance integration with a personal computer.

The *USB 2.0* physical interface is made of sockets and cables and they are available almost in all the personal computers, either Mac, Windows or Linux based.
When the *USB Audio 2.0* protocol is implemented over the *USB 2.0* physical interface it is possible to create a very powerful audio chain.

The *USB Audio specification 2.0* defines multiple formats for audio of which *PCM* standard is only one.
The general “raw data” format on which the *USB Audio 2.0* is based can be used for any kind of data including audio.

The two most popular formats for consumer audio are *PCM* and *DSD*.

- *PCM* is a data flux of frames 16, 24 or 32 bits long.
  The sampling rate referred to the frame is up to 384KHz.
  The *I²S* protocol outlines one specific type of *PCM* digital audio communication with defined parameters and probably it is the most popular.
  Mainly for its popularity, the audio users identify *PCM* and *I²S* as the same thing and this is done in this document too.

- *DSD64* and *DSD128* are data flux of a single bit, at a sample frequency 64 or 128 times the 44.1KHz CD frequency. Often the 48KHz DAT frequency is chosen for the reference instead of the 44.1KHz but the DSD is still called *DSD64* and *DSD128.*
  For the professional audio the formats available are *DSD256* and/or *DSD512* too as well as *DSD64* and *DSD128*.

The implementation of the *USB Audio 2.0* protocol is done by the driver installed into the personal computer and by the firmware installed in the DAC. The firmware is still a software but specialized to manage hardware.
In the *ZeroUno* **DAC** the USB Audio 2.0 capability is based on the XMOS chip X1S.

Onto the Personal Computer, the *USB Audio 2.0* driver let the Personal Computer acts as a powerful audio player. The availability of *PCM* and/or *DSD* depends on the driver installed.

For the Mac personal computers a part of the USB AUDIO 2.0 is native in the Operating System.
For the Windows users the ASIO driver is always needed to use the USB AUDIo 2.0 formats.

However, the latest release of Apple®’s operating system OS 10.x incorporates a *USB Audio 2.0* driver but it supports only *PCM* format.
Furthermore, the central audio engine of OS 10.x, CoreAudio, only supports *PCM* as well.

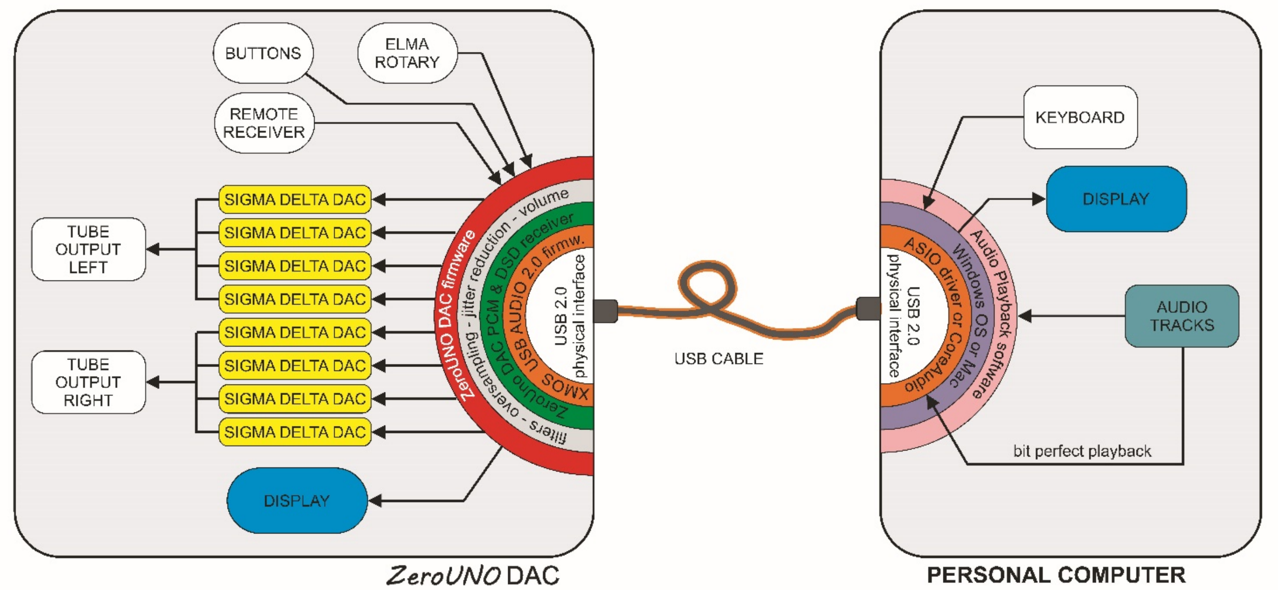
To let the *PCM* format to carry the *DSD* format was defined an open standard: the standardized way to put *DSD64 and DSD128* audio data into *PCM* frames is *DSD over PCM (DoP)*.

When using the Windows platform things are little easier because Windows by nature does not fully support *USB Audio 2.0* and what it does is limited to *PCM* only at a sample rate of 96kHz or less.
So for Windows users always there is the need of a dedicated software to have the *USB Audio 2.0*.
This software is known as the ASIO driver and can be of different level of performances, so it can implement the *PCM* only and/or the native *DSD* only and/or any other *USB Audio 2.0* format.
It is specific of the audio unit interfaced.

As more universal the ASIO driver will be as more sophisticated and expensive will be.

Anyway, having the *DoP* standard ready it is useful for Windows too and not only for Mac OS, simplifying in this way the development of the ASIO drivers for Audio applications cause instead to manage all the possible audio formats it can be written only for *PCM*.

The below schematic summarizes the concepts just described. It is only a graphical representation of the themes just introduced and not a technical scheme.



DSD256 and *DSD512*, when there is a USB interconnection, cannot be managed as *DoP* so the only way to connect an equipment able of *DSD256* and/or *DSD512* is a dedicated custom driver able of *native DSD.*

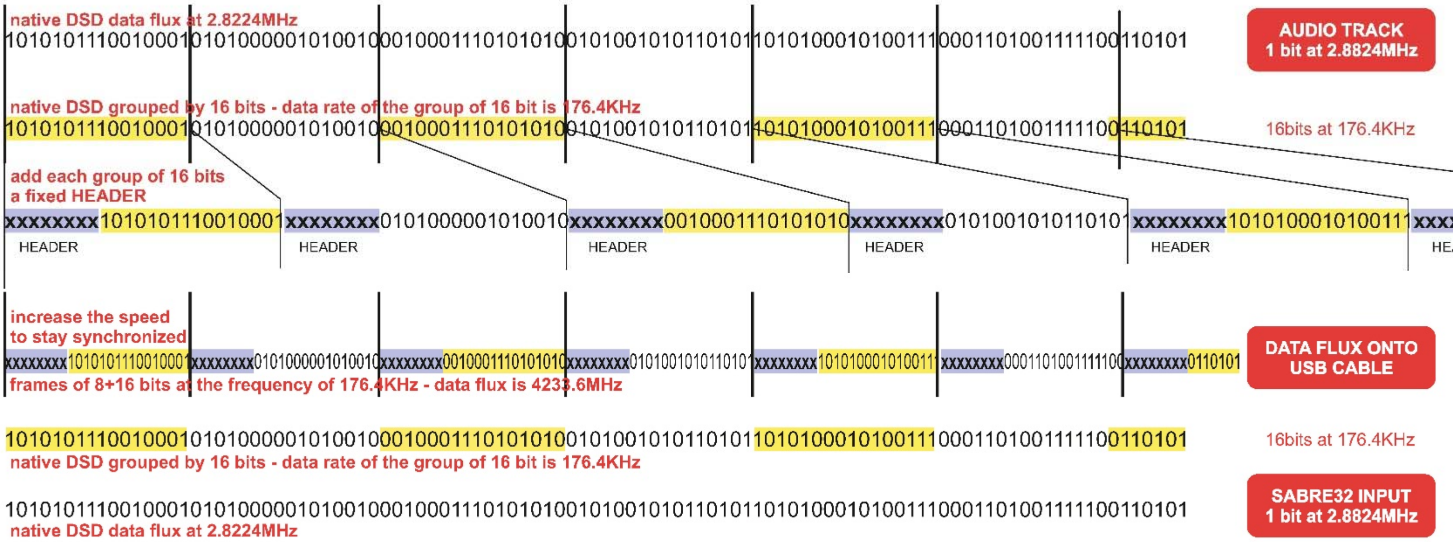
For Windows users it is easier because everything depends on the ASIO driver available; Windows does not limit the performances because totally leaves to the ASIO driver the capability of *DSD256* and *DSD512*.

THE DoP OPEN STANDARD

A *DSD64* has a frame made of 1 bit only and a sample rate of 2.8224MHz.
A *16bit 176.4KHz PCM* frame has a data rate of 16x176.4K = 2.8224Mbit/sec.
And it is exactly the same data rate of the *DSD64*, this is the basic idea of the *DoP*.

If 16 consequential *DSD* bits are packed in one 16 bits *PCM* frame instead of one *PCM* audio sample, the *DSD* flux can be seen as a *PCM* structure. The DSD bits are only packed and not changed.

There is only the need to distinguish a real PCM frame from a *PCM* used to carry a *DSD* group. This is done adding 8 HEADER bits, always the same, to each group of 16 bits.
The by-product is only the need of a higher data rate, 4.2336MHz if compared to the original 2.8224MHz to have a totally transparent transfer, but this 4.2336MHz is a very small portion of the 60MHz band of an USB 2.0 connection.

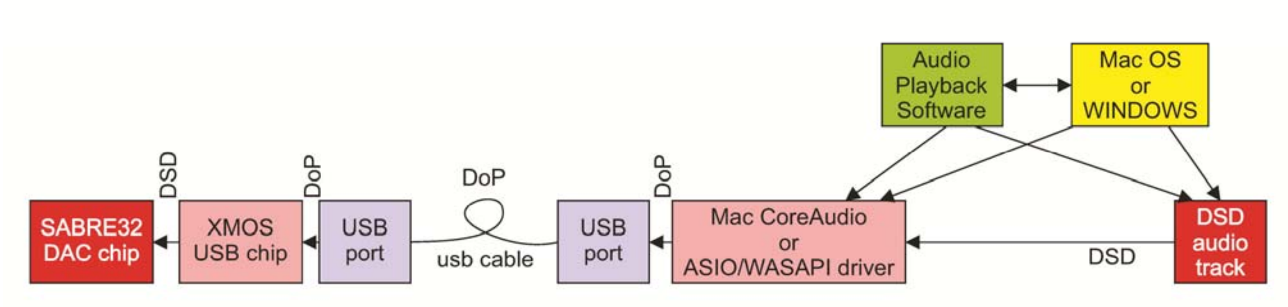


The "driver" (really CoreAudio for Mac OS and ASIO/WASAPI for Windows) has to collect 16 DSD bit, pack into a 24 bit *PCM* frame, and add the HEADER that it is always the same.

The HEADER is a number between 0 and 255.

The receiving USB AUDIO 2.0 unit has to do the reverse job:
- stay tuned on the coming digital signal,
- isolate the DSD HEADER,
- collects 16 consequential bits and remake the original DSD flux.

The DoP does not convert the DSD to PCM but only group the DSD bits in block of 16 and add a marker to distinguish the DSD from the PCM.



The *DoP* is a perfect bit process because the original data are never changed.

Please note the DAC chip works always in *native DSD*.

For double rate DSD, *DSD128* (or sometimes named *2xDSD*), the solution is simple and exactly the same. Still group the bits in frames of 16, add a HEADER but at double sampling rate, 176.4KHz x 2, i.e. 352.8KHz.

The limit of *DoP* is *DSD128* because for *DSD256* should need a sample rate of 176.4KHz x 4 = 705.6KHz and it is over the *PCM* limit of 384KHz.

When the *PCM* 768KHz will become popular, not only because some hardware manufacturers has introduced on the market units able of this sampling rate but mainly because the availability of music tracks / programs rated at 768KHz, the *DoP* will be updated to the *DSD256*.

THE *ZeroUno* DAC SOLUTION

Inside the *ZeroUno* **DAC** the parts involved with the USB Audio 2.0 are the chips SABRE32 ES9018s and XMOS X1S and let the *ZeroUno* **DAC** be native DSD256 ready.

Regarding the Mac OS based personal computer it is possible the maximum integration and all the Audio formats possible with the Mac CoreAudio are available: PCM up to 384KHz and DoP up to DSD128.

Regarding the Windows based personal computer, the *ZeroUno* **DAC** ASIO driver is the official XMOS driver for stereo application, and it is able of PCM up to 384LHz and DoP up to DSD128.
By the ASIO driver the performances of a Windows based personal computer and a Mac OS based personal computer are exactly the same.

When XMOS will release the official ASIO driver for stereo playback able of native DSD up to DSD256, the driver will become a standard part of the *ZeroUno* **DAC.**
Without any hardware or firmware changes, but only upgrading the ASIO driver installed in the Windows based Personal Computer, will be possible to play native DSD. *ZeroUno* **DAC** already played native DSD256 with plain success using a third party ASIO driver.

ASIO, WASAPI, DIRECT SOUND OR KERNEL STREAMING

The *ZeroUno* **DAC** is able of *PCM* audio format and *DSD* audio format through its USB input.

The USB interface is based on the XS1 XMOS chip and the Windows users must install the XMOS driver supplied with the *ZeroUno* **DAC** before to use it.
The driver implements the USB Audio 2.0 format into Windows Operating System (OS).

The Mac OS users has nothing to do because the USB input of the *ZeroUno* **DAC** is fully compliant with the USB Audio 2.0.

The installation of the driver for Windows gives 4 different drivers:

- ASIO
- WASAPI
- Direct Sound
- Kernel Streaming.

With the ASIO driver in a Windows based personal computer there is no difference with the performances of a Mac OS implementation.
Windows+ASIO and Mac OS based implementations fully implement the USB Audio 2.0 standard and make possible the Perfect Bit Playback mode with the *ZeroUno* **DAC**.

The valid Audio formats are PCM up to 384KHz, 16, 24, 32 bits and DSD64 or DSD128 by the DoP open standard.

Windows users can experiment with the WASAPI driver too.
WASAPI (Windows Audio Section App) is the Windows solution for the Perfect Bit Playback Mode compliant with USB Audio 2.0 standard. It is available after Windows Vista.
This driver is an alternative to the ASIO driver.
WASAPI usually it is not so efficient as the proprietary ASIO driver.

Last, Windows users have the Direct Sound and a Kernel Streaming drivers available too.
They correspond to a playback mode available in the Windows OS since Windows XP.
These drivers are known as universal drivers because work in any Windows installation but these two drivers do not perform a bit-perfect reproduction:
All the audio tracks are resampled and managed by the Windows internal mixer (KMixer).

GLOSSARY

USB Audio 2.0	specification of formats for Digital Audio involving with the USB capability of the personal computers
USB 2.0	physical interface available almost in all the personal computers
USB 3.0	physical interface available in the last generation of personal computers
ASIO	driver/firmware for Windows to implement USB Audio 2.0
CoreAudio	Mac Audio engine
PCM	Pulse Code Modulation. Audio format. Each sample is made of 16, 24 or 32 bits. Sampling frequency up to 384KHz. Includes I²S, LJ32, RJ32, RJ24, RJ20, RJ16.
I²S	Audio format. It is of the PCM family.
DSD	Direct Stream Digital. Audio format. One bit signal flux with sampling frequency up to 22.5792 MHz. Each sample define if the samples signal drops or rises.
DSD64	DSD at 2.8224MHz sampling frequency. It is 64 times the basic CD sampling rate of 44.1KHz.
DSD128	DSD at 5.6448MHz sampling frequency. It is 128 times the basic CD sampling rate of 44.1KHz.
DSD256	DSD at 11.2896MHz sampling frequency. It is 256 times the basic CD sampling rate of 44.1KHz.
DSD512	DSD at 22.5792MHz sampling frequency. It is 512 times the basic CD sampling rate of 44.1KHz.
DoP	DSD over PCM. Standard defined by USB Audio 2.0 to pack the DSD64 and DSD128 in a PCM frame.
Latency	The time needed by a software component to be executed.