PRODUCT INFORMATION SHEET

CD Transport - *Blu*

Product Description

The Chord CD Transport has been designed to match, both in performance and style, the highly successful DAC64 and also completes the Chord Choral range. For the first time, customers can complete their Chord system with a truly stunning source component. Called **Blu** the CD Transport will share the same ground breaking technology that has made the DAC64 one of the most successful Digital to Analogue Converters ever produced. As a dedicated state-of-the-art, 2-channel CD Transport, Blu will take full advantage of the WTA Filter technology used in the DAC by having it's own 4096 tap length WTA filter.

With features like selectable up sampling from 44.1 to 176kHz and a dedicated word clock input, **Blu** can be also used successfully outside a dedicated Chord system. The digital connections on offer are; 2 x AES/EBU and 2 x BNC for dual data output, 1 x BNC & Toslink optical for standard data output (all selectable via a toggle switch), 1 x BNC input for word clocking synchronisation, selectable output sampling frequencies: 44.1kHz, 88.2kHz and 176kHz.

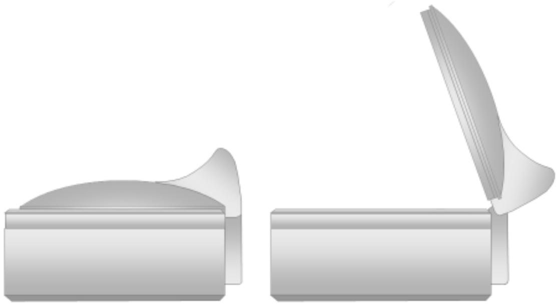
Direct Access Functions on CD Transport

- Play
- Stop
- Previous
- Next
- Pause
- Skip Forward >>
- Skip Back <<
- Program
- Time
- Scan
- A/B
- Fast
- Standby
- Random
- Track numbers 1 - 10

Product Specification

Back Panel Connections

- 1 x TOSlink optical – digital output
- 1 x 75ohm SP/DIF BNC – digital output
- 2 x SP/DIF BNC for dual data digital output (left op 37.5ohm & right op 37.5ohm)
- 2 x AES/EBU Balanced XLR for dual data digital output (left op 52.5 ohm & right op 52.5ohm)
- Single toggle switch for selection of dither on / off
- Single toggle switch for selection of 44.1kHz, 88.2kHz & 176.4kHz
- 1 x BNC for Word Clock Synchronisation



Other Features

- Chord Electronics high-frequency switching power supply: automatic input voltage selection with PFC
- Dedicated CD only laser mechanism for uncompromised music replay
- Can be used in Choral rack and also flat on a normal shelf
- Casework machined from solid billet aluminium
- All functions have direct access via buttons on the CD transport or via full Chord Touch screen remote control
- Fully upgradeable via EPROM – easily accessed on the bottom of the unit.
- Self supporting lid design that automatically stops play when lifted and also holds the disc puck in perfect place
- The dimensions of the CD transport are the same in width and depth as a DAC64 (excluding CD lid)
- A two shelf Choral rack will be available for those who want to use it with the DAC4 as a stand alone CD player

Functionality:

Dual Data Switch set to OFF

	Frequency switch Position		
	44.1kHz	**88.2kHz**	**176.4kHz**
Optical	44.1	88.2	88.2
Single BNC	44.1	88.2	88.2
Dual BNC top	44.1	88.2	88.2
Dual BNC bottom	44.1	88.2	88.2
Dual XLR top	44.1	88.2	88.2
Dual XLR bottom	44.1	88.2	88.2

Dual Data Switch set to ON

	Frequency switch Position		
	44.1kHz	**88.2kHz**	**176.4kHz**
Optical	44.1	88.2	88.2
Single BNC	44.1	88.2	88.2
Dual BNC top	44.1	88.2 left	176.4 left
Dual BNC bottom	44.1	88.2 right	176.4 right
Dual XLR top	44.1	88.2 left	176.4 left
Dual XLR bottom	44.1	88.2 right	176.4 right

4096 WTA Filter

The WTA filter algorithm has taken twenty years of research to develop. It solves the question as to why higher sampling rates sound better. It is well known that 96 kHz (DVD Audio) recordings sound better than 44.1 kHz (CD) recordings. Most people believe that this is due to the presence of ultrasonic information being audible even though the best human hearing is limited to 20kHz. What is not well known is that 768 kHz recordings sound better than 384 kHz and that the sound quality limit for sampling lies in the MHz region. 768 kHz recordings cannot sound better because of information above 200 kHz being important – simply because musical instruments, microphones, amplifiers and loudspeakers do not work at these frequencies nor can we hear them. So if it is not the extra bandwidth that is important, why do higher sampling rates sound better?

The answer is not being able to hear inaudible supersonic information, but the ability to hear the timing of transients more clearly. It has long been known that the human ear and brain can detect differences in the phase of sound between the ears to the order of microseconds. This timing difference between the ears is used for localising high frequency sound. Since transients can be detected down to microseconds, the recording system needs to be able to resolve timing of one microsecond. A sampling rate of 1 MHz is needed to achieve this!

However, 44.1 kHz sampling can be capable of accurately resolving transients by the use of digital filtering. Digital filtering can go some way towards improving resolution without the need for higher sampling rates. However in order to do this the filters need to have infinite long tap lengths. Currently all reconstruction filters have relatively short tap lengths – the largest commercial device is only about 256 taps. It is due to this short tap length and the filter algorithm employed that generates the transient timing errors. These errors turned out to be very audible. Going from 256 taps to 1024 taps gave a massive improvement in sound quality – much smoother, more focused sound quality, with an incredibly deep and precise sound stage.

The initial experiments used variations on existing filter algorithms. Going from 1024 taps to 2048 taps gave a very big improvement in sound quality, and it was implying that almost infinite tap length filters were needed for the ultimate sound quality. At this stage, a new type of algorithm was developed – the WTA filter. This was designed to minimise transient timing errors from the outset, thereby reducing the need for extremely long tap lengths. The WTA algorithm was a success – a 256 tap WTA filter sounded better than all other conventional filters, even with 1024 taps. WTA filters still benefit from long tap lengths; there is a large difference going from 256 taps to 1024 taps.

Currently the DAC64 uses 1024 taps. The filters are implemented in FPGAs (Field Programmable Gate Arrays) using a specially designed 64-bit DSP (Digital Signal Processing) core. The Chord CD transport will use a 4096 taps WTA filter. This will further improve the ability of the filter to reconstruct transient timing. We know this to be true from the DAC64, which already increased the tap lengths used from 516 to 1024. What this will mean in terms of sonic performance is when combined with the DAC64 we can achieve perceived improvements in bass, rhythm, timing and sound staging.

All of the above innovations are implemented in Xilinx Spartan series FPGA's. These FPGA's can offer in the CD transport 400,000 gates per device, and merely updating the EPROM memory chip can easily change the design, thus future proofing is assured.

Sampling Frequencies

As already explained above it is well known that the higher the sampling frequency, the better the soundstage quality i.e. 96kHz recordings (DVD-A) sound better than 44.1 kHz (CD) recordings.
However we at Chord are using up-sampling correctly, the CD Transport will up-sample the signal before transferring it into the Digital to Analogue converter by whole factors, not derived clock signals i.e. 96kHz or 192kHz. We are outputting from the transport, true multiples of the original sound recording frequencies:

44.1kHz x 2 = 88.2kHz
88.2kHz x 2 = 176.4kHz

We have specifically chosen these frequencies because they do not risk adding any digital noise into the signal that 96kHz and 192kHz can suffer from because they are not truly up-sampled by whole factors.