Back to the Origin ; Back to the Future – Audio Note DAC-5

Introduction

April 1981, two hundred reporters from all over the world gathered at Salzburg, attended the CD show ceremony demonstrated by Karajan. Whirling a silvery and shiny Polygram CD in his little finger, in high pitch Karajan said: All else is gas light !

From then on, LP was put onto the stake, succeeded by the sonic deficient CD. According to the Philips/Sony pre-defined standard, CDs are digitized signal with a word length of 16bits sampling at 44.1KHz. 16 bit was set based on the assumption that 1 bit could have 6dB of headroom thus 16 bits could offer 96dB of dynamic range. The employment of 44.1KHz sampling frequency was even an obstinate decision based on the classical Shannon's theorem. According to the researcher's application of this fundamental theorem, a musical signal can be recovered completely by a sampling frequency two times of the original signal frequency. The limit of human sonic frequency was 20KHz. Adding a bit of 'bonus' and times two made up to 44.1KHz. In other words the upper frequency limit of CD is 20.5KHz.

Oversampling excels without oversampling

People soon knew that the CD's virtual 96dB dynamic range and 20.5KHz upper frequency limit was indeed a big step backward when comparing to the analog recording performance by 1981. Digital-to-analog conversion must be chopped at 20.5KHz that below 20.5Khz was analog signal while above 20.5KHz would still be digital signal. Scientist believed that it is essential to prevent digital pulses from leaking into the analog path or else explosive noise could be heard. Filter with very steep(or theoretically infinite) skirt is called brick-wall filter. In the early Japanese D/A converters the filters were built on analog hardware. The requirement was stringent - it must attain –50dB attenuation at 24Khz. The result was tremendous distortion that the treble region was awfully reproduced. The workaround was the oversampling technique suggested by Philips. The initial oversampling technique was using 4 times sampling frequency, that is, 176.4KHz, and then passed through digital filtering. Listening test proved that oversampling digital filtering was far better than non-oversampling analog filtering. The key factor is not only digital filter has better response than analog filter but more important being that for non-oversampling's case to achieve –50dB attenuation at 24KHz the cut-off frequency is only 3.5Khz apart from the CD's upper frequency limit. On the other hand 4X oversampling(2x24K=96K) could push the frequency distance(96K - 4x20.5) to be 14KHz apart.

Thus oversampling had made a great progress from standard sampling rate. At that time other companies like Marantz, B&O, Revox followed to implement oversampling into their CD players. But don't forget that resolution was only 14bits. 16bits wouldn't be put into market till Philips had sold out the stock of their 14bits chips.

Noise shaping was another idea raised by Philips. During the analog-to-digital conversion process, digital noise was distributed evenly across the whole audio band. Thus after 4x oversampling only 25% of noise fell back into the audio frequency. This is equivalent to gaining 13dB of extra dynamic range.

The advantage of oversampling was well-known to everyone at that time. However it was indeed a digital variation of companding technique which had been invented by Dolby back several years ago to improve signal-to-noise ratio. Thus noise shaping was also nothing new but a free lunch deriving from oversampling.

Another thing in place is jitter. Digital experts had found that jitter could be reduced drastically through oversampling. Though jitter was infamous among people, not many of us aware that the cause of jitter could be electronic means or mechanical means. Electronic jitter is the error from DSP manipulation that could be effectively reduced by oversampling. Mechanical jitter on the other hand was due to CD drive instability, laser head misaligned or disc bending that cause the data rate fluctuates. Unfortunately Oversampling could not cure mechanical jitter.

1972, digital recording began to develop with word length of 14bits. By 1979, all major recording companies were using digital recording. The famous machine at the time was Soundstream 16bit. 3M multichannel, Karajan's anointed running horse had its time too. But the most popular one was Sony 1600 series. The sampling frequency prevailing in professional circle at that time was 48KHz. While discontenting with the performance of 16/44.1K, Decca developed its own 18/48K standard for recording. So the announcing of CD playback standard at 1979 by Philips was incompatible to the prevailing digital recording standard that frequency conversion was required. Technically the conversion of 16 bits or 18 bits of data stream from 48KHz data rate into 44.1KHz was a tough task. Just to imagine that processing unit receives 18 bits of 'one' or 'zero' at a speed of 48,000 bits per second and this streaming data have to be converted into 16 bits of 'one' and 'zero' at real time with no delay plus the error rate must be less than +/- 4 bits per second !

Processor manipulation of oversampling is mainly the algorithm of multiplication followed by division. This is relatively much simpler than 48K/44.1K frequency conversion. Thus by 70s the already-mature conversion technique, plus noise shaping from oversampling were something we have to be proud of.

Nowadays lots of recordings have employed similar principle to record signal at high bit rate like 20bits/96K, convert to 16/44.1 and play around with specific algorithm to spread the noise so as to push the majority of noise out of 22KHz. These CDs have proved to improve the sonic behavior a great deal. Whatever we name them, 'bit image processing', 'bit mapping' etc are all originated from noise shaping principle.

Reprocessed CDs are better than their old recordings most of the time. New recordings have even excellent performance that is sometimes hard to believe CD just has 16 bit. The main reason being the digital noise of new recordings is further reduced, which implies a pure sound. Putting it straight, the 20 bits impossible mission the scientists have achieved was in fact a 'noiseless 16bit' signal.

Finding more and more DA converters trying to get to increase the number of bits on their decoding engine, some smart audiophile experts saw something differently and went, as all CD right now have been processed with oversampling, why don't we just built a DA converter without oversampling that could just do the job right ?

Analog filtering by transformer in 1991

Way back to 1991, Peter Qvortrup of Audio Note have successfully used transformers to do analog filtering and got the patents in UK, US, Germany, Australia and some other countries. By 1995, 1Xsampling DAC with analog filtering from Audio Note have emerged in the market. 2000, Audio Note DAC-5 is launched. DAC-5 is not the first non-oversampling, analog filtering DAC in the marketing. Back to 1981 this kind of DACs were everywhere. But DAC-5 was the first 1Xsampling DAC putting transformer in between digital-to-analog output and analog filtering.

It was told that to acquire the brick-wall effect analog filter (–50dB attenuation at 24KHz), phase distortion would be horrible. If it was not a brick wall leakage of digital signal above 24KHz into the audio band would happen. Thus the stronger the brick wall the more serious the phase distortion.

Peter's secret weapon is the *interface transformer* in between the DAC and analog filtering stage. Would analog filter serve well enough to block the digital? Peter said, 'Don't worry! A minor leakage of digital into analog couldn't be heard by human ear.'

Audio Note is good to argue that their products have 'high distortion but excellent sound'. Peter said, 'Power amplifiers from Audio Note could have 8% of THD. But what, we have the best sound!' How much is the THD of Audio Note DAC-5? God knows. But in terms of sound it is hard to find an opponent. The launching of DAC-5 is another confrontation of Audio Note against the measurement scientists in the Hi-End world.

From input signal to Analog Devices 18bits AD1862N DA chip, the shortest path is taken. The pure 16bits data from this chip is then fed to the interface transformer to carry out the first stage of filtering. The signal is then fed to the second stage 3^{rd}-order 18 dB RC filtering consisting of silver resistors and capacitors. The resulting output signal is still an audio with quite an amount of digital residue signal above 22KHz that one can easily observe on the oscilloscope. But Peter guaranteed that this residue couldn't be heard and would do no impairment to music listening. The linear stage is similar to the M10 pre-amplifier. Line output passes through output transformer, offering both balance and unbalance output.

The power supply of DAC-5 is no doubt a huge one, that occupies 2/3 of the whole space. It is a shunt regulator composing of the best components.

Why analog filtering ?

Peter Qvortrup has raised a big accusation against digital filtering, that would be so controversial to cause a lots of strikes back from the digital camp. As audiophiles we just stay tune to see what's next.

Peter analogous digital filtering to the proven zero-feedback theory. The failure of negative feedback on music signal, was because it assumed the time is frozen that the feedback signal could do correction to the original signal. Unfortunately the feedback signal was always a delayed one. Zero-feedback is the real man in the show. Similar to feedback mechanism, in order to attain brick wall response, digital filtering employed sinc function(that is, sin(x)/x) mode of operation to minimize phase distortion. However, sinc function takes quite some time to complete the whole impulse response. Although only a small portion of the sinc pulse is required in filtering, the time delay is still in the order of several milliseconds. This minimal delay of time caused *smearing* effect in digital filtering. Peter said by AB comparison one can easily heard the smearing due to feedback. I have put some time to do in-depth studies of the audible drawback of feedback since I the zero-feedback M10 preamplifier became my favorite control unit. To put it straight, smearing is a delay in sound wave, such that the front wave hits the back wave. The result is a deterioration in the sound picture, restricted soundstaging, mechanistic rhythm and liveless music. Prolonged hearing of smearing amplifier would get used to it with no sense of negative feedback distortion. Some audiophiles on the other hand tried to prevent from hearing zero-feedback to preserve their self-respect and pockets.

If the accusation of Peter on digital filtering is valid, foundation of digital technology would be shaken.

Listening

Putting the technology stuff aside, let's go to the listening. *Ray here now announce, Audio Note DAC-5 is one of the very best DAC that one can get in the market.*

DAC-5 is also the first tube DAC Ray put into his Hi-Fi room for gear comparison. The main reason being the price – a list price of 18,500 pounds.

All the factors below could be the reason(s) to make DAC-5 a great sounding box.

DAC-5's features are :

1. 1X sampling, that is, 44.1KHz ;
2. Interface transformer in between DAC output and analog filtering, taking up the first stage analog filtering task ;
3. Twin tube(2 5687 triode) anode output, zero-feedback ;
4. 18dB 3rd order analog filtering ;
5. transformer coupling at output for unbalance and balance output.

Dynamic, Energetic and Expeditious sound

In my opinion, the behavior of DAC-5 zero-feedback/analog filtering has something in common to zero-feedback amplifiers but there are also dissimilarities. The first time I heard on all-tube CD/control/amplifier system, the very feeling of the sound is 'fast', even faster than any solid-state DAC. If it were happened on LP system, such prominent behavior would make one thought the turntable speed is off tuned from 331/3. For CD drive, speed is obviously out of question, while tones are also at the right pitch. Thus this fast feeling was extraordinarily mystical. People used to describe crispy or muscular sound as 'fast'. But the real fast sound should be smooth and pure. The 'fast' of DAC-5 make the sound picture more transparent, the relative timing of reverberation and mother sound is crystal clear, residues at the top end extends beyond imagination. The dynamic of base and near-base is aggressive and possessing, pushing to the excitement limit. Believe it or not, the sound of 'All-Tube' CD system has mere of tube sound, even closer to solid-state sound, the zero-feedback solid-state sound, which entirely make me indulge into it.

The supreme character of M10 century preamplifier is its 'fast' and mere tube sound. On working with DAC-5, they are a prefect match, supplementing each other. M-10 was the beloved baby of Peter Qvortrup that he was proud of. I loved this preamplifier for its passion, even to the degree of sweaty violence. The twin 5687s of DAC-5 have similar character.

The tonal balance of DAC-5 is excellent. There is no emphasis in the low, middle or top end. The sweet coloration of low cost tube gears was not ever added in. Obviously Peter is reluctant to do so. Audio Note has gone so far beyond that the opponents are hard to reach.

It seems that the next evaluation step would be to jog a number of CD codes, their track number etc. Not this time. The reader should have known the result.

Gear-to-gear comparison of DAC-5 with other top-end DAC would mainly be a matter of taste. If you are looking for dynamic, energetic and expeditious sound, DAC-5 is your cup of tea.

It's worth to mention that DAC-5 is so faithful to the record that on good recordings it sounds great while on poor records it gives you awful sound. The line output is inverting phase. Balance output outperforms unbalance out. While using XLR one has to aware of the pin out of ground of the transport. Some transport's(e.g. Mark Levinson) XLR output could trigger the protection circuit of DAC-5. Separating the AC connection of the transport and the DA could fix the problem. Unbalance connection on the other hand wouldn't cause the trouble. Unfortunately my unbalance digital cable was not the top-end to match DAC-5 that I am obliged to use XLR.(The XLR input of DAC-5 is not differential but single-ended.) But I have found a trick here. While XLR and RCA are both in used not only protection circuit is not triggered the sound is also cleaner than sole XLR connection.
Beard Peter cannot give a reason either. But since bi-wiring speaker cable always excels single wiring, why bother to bi-wire digital cable ?