SECRETS of Home Theater and High Fidelity

Product Review - Perpetual Technologies P-1A Digital Correction Engine and P-3A DAC - September, 2000

Stacey Spears

Perpetual Technologies P-1A Digital Correction Engine

Frequency Upsampling**:** 96 kHz Standard, 192 kHz Option

Word Length Upsampling**:** 18-bit, 20-bit, 24-bit

Digital Inputs**:** Coax RCA, I^2S, AES/EBU

Digital Outputs**:** Coax RCA, I^2S, AES/EBU

MSRP**:** $950 USA, $399 Additional for Speaker Correction Software, $699 Additional for Room Correction Software, 192 kHz Sampling Upgrade Price To Be Determined.

Perpetual Technologies P-3A DAC

Capable of Decoding up to 96 kHz, 24-Bit Signals, 192 kHz Option

Digital Inputs**:** Coax RCA, I^2S, Optical Toslink, AES/EBU

Analog Outputs**:** RCA

MSRP**:** $699 USA

Introduction

Perpetual Technologies (PT) is one of Mark Schifter's new companies. As you know, he founded Audio Alchemy some years ago, and then moved on to do other things. After spending some time with Genesis, he founded PT. It is a lot like Audio Alchemy, in that the products are state of the art, but without the price. Peter Madnick, also of Audio Alchemy, designs for Mark at PT. They don't have milled chassis and fancy readouts. All the money is on the inside. If you have had Audio Alchemy in the past, I am not telling you anything new. If not, you are in for a pleasant surprise. The P-1A and P-3A are PT's first products, and we have to say, "WOW!" They upsample, enhance, and adjust for your speakers' as well as your room's deficiencies, then decode, all at leading edge levels.

P-1A

Front Panel

The front panel has two buttons, one on the top, which is the input selection, and one on the bottom, which is used to program the 1A. It also has 7 LEDs broken out into two sections 1-3 and 4-7. Each LED is capable of three different colors. These are used to determine the state of the 1A.

LEDs 1-3, going from the top of the 1A down, represent the input. 1 = I^2S, 2 = AES/EBU, and 3 = Coax. LEDs 4-7 are used to program the 1A. Here, you select bypass, resolution enhancement, speaker/room correction, the sampling rate (44.1, 48, 96, and 192 (an option), and the output word length (16, 18, 20, and 24-bit).

Back Panel

Inputs

- I^2S (5-pin mini-DIN)
- Balanced (AES/EBU)
- Unbalanced (Coax (S/PDIF)
- USB
- 12V DC

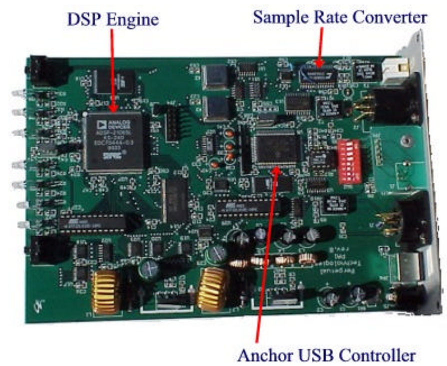
I find it a bit unusual that it is missing an optical input (Toslink), but there is only so much room on the panel. There needs to be a BNC too, at some point.

Outputs

- I^2S (5-pin mini-DIN)
- Balanced (AES/EBU)
- Unbalanced (Coax (S/PDIF))

Under the Covers

The 1A consists of three main subsystems (see photo at right), an Anchor USB micro controller, the Analog Devices 21065L SHARC DSP engine, and the Crystal CS8420 sample rate conversion (SRC) chip.



The Anchor is responsible for controlling the operation of the 1A. This includes front panel buttons, displaying LED status, and interacting with the SHARC and SRC chips. The SHARC performs all of the resolution enhancement, word-length adjustment (re-dithering), and speaker/room correction. The SRC, as its name implies, converts from one sampling rate to another.

The 1A can transcode from one format to another. For example, you can use the I^2S input and output S/PDIF, or vise versa. All internal processing is actually done in the I^2S format. If you use the I^2S input (meaning an I^2S connection from the transport to the 1A), then you cannot upsample the signal. The reason is that the SRC can only handle up to 96 kHz. If you want to pass a 192 kHz signal through the 1A you must bypass the SRC. When you bypass the SRC, you still get all of the other 1A features.

All of the software or algorithms are stored in flash memory (EEPROM). The flash is updated by using the USB port and your PC. You will be able to download new software from the Perpetual Technologies website, including speaker and room correction and then transfer the software to the 1A with your PC. This is the first hi-fi product that I am aware of that offers a USB port.

The 1A also has some very serious power supplies! The 1A comes with a wall-wart adapter, but you can add an optional beefier outboard supply. The 1A board also has a ground plane, which is necessary to keep the noise low. Some of you EEs might be saying, "Ground plane, don't they all have one?" Duh! Well, let me tell you that not everyone uses a ground plane. The power supplies on the 1A are so good, that if you plugged it into mud (providing mud could deliver AC), it would be able to completely filter out the garbage!

What exactly does a sample rate converter do? By increasing the sampling rate (upsampling), it helps out the DACs reconstruction filters later on. If you select 96 kHz as the output sample rate, then all incoming sources, typically a 44.1 kHz signal, are converted to 96 kHz. Upsampling from a 48 kHz source to 96 kHz is a direct integer upsample, i.e., it creates new samples in-between the original samples. This will be the case with PCM data on DVD, DAT, and DSS. For CDs, which are 44.1 kHz, it is performing non-integer-based upsampling. What that means is that it is no longer a direct integer of the original sample rate. This requires that the interpolation algorithms create all new samples. Non-integer-based upsampling requires a lot of DSP horsepower to do it correctly.

I wish that the P-1A and P-3A offered the ability to perform direct integer sample rate conversion on all incoming sources (output 2x the sample rate). You would get 96 kHz from 48 kHz and 88.2 kHz from 44.1 kHz. With that said, I did prefer the sound of the 96 kHz to the original 44.1 kHz. *(Note from JJ: I spoke with Mark Schifter a few weeks ago about this, and they plan to offer integer upsampling software if you want 88.2 kHz instead of 96 kHz.)*

User Interface

The UI of the 1A is very Audio Alchemy, and programming of the 1A is a four-step process. You first select the input, and then the program, sample rate, and finally the output word-length. I am not really sure you could actually program it if you did not see the manual. You could get it to work alright, but you would not know what it was doing. It takes a little fiddling, but once you understand the buttons, it is fine.

During and after programming, you have to pay attention to the LEDs that are active and their color. You also have to press both buttons at the same time for part of the programming.

Scenarios

There are various applications for the 1A. One of the most common uses is to insert it between your CD/DVD player and your DAC. This is a very practical application for a high quality two-channel music system. The output of the DAC would then be fed into your analog preamplifier.

Another use might be in between your CD/DVD player and your digital processor. You will get the same benefit as the two-channel system. In this situation, resolution enhancement, world-length adjustment, and sample rate conversion need to be bypassed if you are listening to a compressed format like DD or DTS. I am not sure what happens if you have the speaker or room correction option since they do not have the coefficients for my speakers yet. I am assuming that in order to do speaker/room correction on a compressed format, the signal would have to be decoded and then re-encoded in real-time by the 1A.

If your home theater surround processor has digital outputs, then you could insert three 1As after the sources (DD or DTS) have been decoded on the digital outputs. In this scenario you could take advantage of everything. You would need three DACs to go after the 1As. (The P-3As would work nicely here.) But then you have to worry about volume control. If you let the processor handle that it would have to be done in the digital domain and the digital output would have to be variable. Or you would need to use an external volume control. You would plug the three DACs into this six-channel volume control and be ready to go. It's a little more work, but may well be worth the trouble if the results are as good when just using the 1A in a two-channel system.

But wait, I have more applications for the 1A including one that I tried. I have a LinxOne digital card in my PC at work. It's a 96/24 digital sound card, and I fed the output of that into the P-1A. Genelec markets digital 96/24 near field monitors that make great PC speakers. You can feed the output of the 1A directly into the speakers. (They require AES/EBU, and it just so happens the 1A is so equipped.)

P-3A

Front Panel

The front panel is virtually identical to the 1A. In fact, the only differences from the two are the stenciled name and that the input and program buttons are reversed. The front panel has two buttons, one on the top, which is used to program the 3A, and one on the bottom, which is used to select the input. It also has 7 LEDs broken out into two sections 1-3 and 4-7. Each LED is also capable of three different colors. These are used to determine the state of the 3A.

LEDs 1-3 going from the top of the 3A down represent the Phase and Sampling rate. LEDs 4-7 are used to select the input of the 3A. 1 = Toslink, 2 = AES/EBU, 3 = Coax, and 4 = I^2S.

Inputs

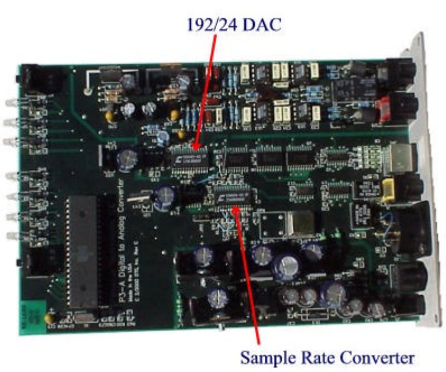
- Optical (Toslink)
- Balanced (AES/EBU)
- Unbalanced (Coax (S/PDIF))
- I^2S (5-pin mini-DIN)
- 9VAC

Outputs

- Unbalanced Analog (RCA) Jacks

Under the Covers

The 3A uses the latest Crystal CS4397 192 kHz 24-bit DAC. The 3A, like the 1A, also uses the Crystal CS8420 SRC chip. All low sample rates signals like 44.1 and 48 kHz are upsampled to 96 kHz.



You can see (photo at right) that the DAC chip itself is only a very small portion of the product. There really is much more to a DAC than just the DAC chip. People often make the assumption that all the DACs that use the same chip will all sound the same.

And just like the 1A, the power supply section of the 3A board is VERY good. A ground plane is also used.

User Interface

Programming of the 3A is more straightforward than the 1A, but then again it's a DAC and does not need all of the wiz bang features of the 1A. You simply select the input you want with the input button, and the program button can be used to invert the phase.

If you want to pass a 192 kHz source, then you have to do a little button combo. Currently, it will not accept a 192 kHz source, but a future upgrade will enable that.

General Use

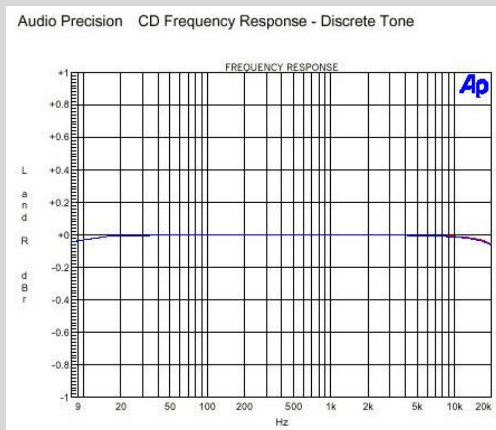
As you will read later when John and Paul comment, the new 1A and 3A are splendid products. The 1A packs many, many features into a single box. It essentially picks up where the Audio Alchemy DTI Pro-32 left off. The 3A actually continues where the Monster Cable Number Cruncher DAC left off.

I spend a lot of time at work, and since I grew tired of swapping CDs in and out of my ROM drive, I have switched over to MP3s to keep me entertained. (It makes me really appreciate the sound of CDs when I listen to them at home.) The resolution enhancement built into the 1A can't cure all the ills of MP3s, but it does go along way to remove the glare that is a result of throwing away most of the data. The upsampling to 96 kHz and high quality DACs found in the P-3A also help a great deal too!

I have had the 1A and 3A for quite some time. We were shipped a couple of pre-production units to use in our DVD Benchmark. What we did was compare the analog outputs of a DVD player to the digital output feeding a high-quality DAC like the 3A. The differences were very startling in both listening tests and measurements.

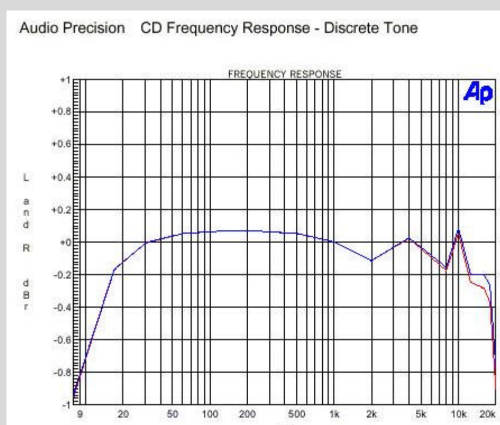
Using the Apex A600D DVD player as a CD player was frightening. It produces some of the poorest quality music that I have ever heard. I get better sound quality from my portable Radio Shack CD player that is several years old. Plugging the 3A into the digital output of the Apex transforms the music wrecker into something very enjoyable. The noise floor alone dropped several dB revealing small nuances in music that were not there when listening to the analog outputs of that player with no 1A and 3A in the path.

Frequency Response on Digital output of Apex, using P-3A DAC



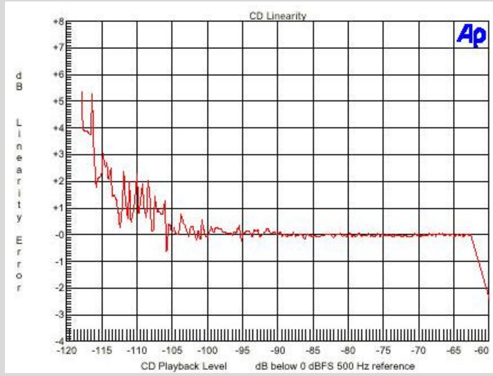
Notice how the frequency response stays close to 0 dB throughout the audible range. Red and blue lines in the graphs represent left and right channels.

Frequency Response using Analog outputs of the Apex player (no P-1A or P-3A)



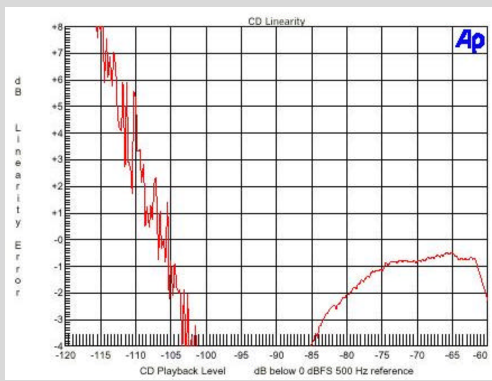
Using the analog outputs of the Apex player, and no P-1A or P-3A, the frequency response is terrible.

P-3A DAC Linearity with Digital output of Apex



The P-3A, when using the digital output of the Apex, shows DAC linearity easily to the -96 dB limit of 16 bits, and actually remains linear below that limit, although the graph eventually rises due to the content added by the noise floor caused by the desirable presence of dither, which allows encoding greater than 16 bit resolution onto a 16 bit system at the expense of a higher theoretical noise level than, say, a 20 or 24 bit system.

Apex DAC Linearity on Analog outputs (no P-1A or P-3A)



To be fair, the plot here is offset by roughly -0.7 dB, but you can still see that even after adjusting the plot to match 0 dB with where the chart starts recording (at about -62 dB), the DACs on the Apex only remain linear to about -75 dB, falling far short of the resolution capable with even standard CD sources, akin to the medium of the vinyl LP. The plot falls absolutely off the chart between -100 dB and - 105 dB, and only rises back due to the noise floor.

I could continue to show you the before and after graphs all day long as to just how good the P-3A is. Colin Miller will actually cover a lot more about the P-3A as the DVD player data are published. We looked a the P-3A on several DVD players, and in 99% of the cases, it made a substantial improvement.

As of this writing we have not had an opportunity to try the speaker or room correction software. Colin Miller and myself did get to listen to the speaker correction at CES 2000. While I can't fully comment on its quality, I do remember the effects were not subtle! When they first turned on speaker correction, I really did not notice any change. But, once they then turned off speaker correction, it was like a slap in the face. I was left wondering what happened. At some point in the future I am sure one of us will get a chance to fully experience the speaker correction software.

Monolithic Power Plant

Both the P-1A and P-3A come with an outboard power supply (wall warts). The Monolithic Power Plant (PP), borrowing its name from the PS Audio power plant, is a 3rd party add-on power supply that was designed to be used exclusively with the 1A and 3A. It does not regenerate the AC like the PS Audio stuff, but it's a beefier power supply than what comes with the PTs. It has two transformers inside, a large EMI filter, a couple of caps, and a full wave rectifier. The PP has one 9V AC output and one 12V DC output. Be very careful to make sure you plug the correct cables into the P-1A and P-3A. Accidentally getting them mixed up could damage the PTs. The Perpetual Power Plant for the P-1A and P-3A is $349, and is available through Monolithic Sound, Inc., 515 Sandydale Road, Nipomo, California 93444.

Conclusion

The P-1A and P-3A represent the next evolution in digital electronics. They offer exceptional performance at a very low price in the a world where the Internet stock craze has allowed companies to charge more because people now have more to spend. The future potential of the P-1A really has me excited. and I can't wait to experience full speaker and room correction.

John Johnson and Paul Knutson will give you a lot more detail on how they actually sound. John's notes are given below. Paul Knutson's comments will be added next week.

- Stacey Spears -

Notes by the Editor - JEJ

Although the P-1A and P-3A (PTs) have lots of ways that they can be configured, I found the choice to be easy, and the configuration procedure to be without any problems. I also had the optional Monolithic Power Supply that has connections for both the P-1A and P-3A, and will deliver more current than the stock wall wart supplies. I recommend getting it when you buy the PTs. After listening to the various choices, I decided on having the P-1A set to 96/24 upsampling, along with enhancement, and the DAC automatically set itself for 96/24 decoding. I used a Nordost Quatrofil digital cable between my McCormack CD transport and the P-1A, and then an I^2S cable (it came with the P-1A) between the P-1A and P-3A. A pair of Nordost Red Dawn interconnects ran between the RCA outputs of the P-3A to a Balanced Audio Technology VK-5i preamplifier. Nordost Balanced Red Dawn connected the VK-5i to a Balanced Audio Technology VK-500 power amplifier, which drove Carver Platinum Mark IV ribbon speakers, Threshold ES-500 electrostatic speakers, or Osborn Eclipse cone speakers (Nordost Red Dawn speaker cables).

The P-3A is an excellent DAC all by itself, but it is the combination of the P-1A and P-3A that made my jaw drop. Together, they delivered some of the finest digital audio I have ever heard. CDs have been notorious for having a slightly harsh edge to them, and this problem was now completely gone. I was able to turn the volume up louder without the slightest hint of irritation. Some music demands loud playing, such as Copland's "Fanfare to the Common Man". In the past, I have used this disc (Telarc) to test many things, and it is a very demanding piece of music, especially the way it was recorded by Telarc. I have to play it loud for the tests, and it has always irritated my hearing a little at these high levels, because that is the way many CDs are. But with the PTs, I enjoyed the music 100% regardless of the volume. I could not get it to a loudness that bothered me from the edgy point of view. The only thing keeping me from cranking it all the way, was the fact that I want to keep my eardrums intact for a while longer.

Cymbals and gongs have an enormous amount of overtones. The Copland Fanfare uses a huge gong throughout the composition. To my complete surprise, not only was the gong much easier (no edginess) for my hearing to take, but it had more substance . . . more body. It was more musical. I have to say that the PT experience is as close to having all the good qualities of our classic analog LPs as I can imagine. I say good qualities, because the PT's sound lacks the pops, hisses, and clicks that plague phonograph records.

OK, the question is, "What the heck is going on here?" Upsampling means that you take the 44,100 samples per second and increase them to 96,000. This requires a lot of mathematics, and "interpolation" is used. This means that you look at a series of samples and try to interpolate (calculate with an intelligent guess) what the curve would look like in between those samples, and generate the additional samples accordingly. Oversampling is a different thing, but related, sort of. Oversampling creates new numbers too, but it is for the purpose of making it easier for the filters to do their job. With oversampling, the new numbers can just be a string of 0s. Upsampling tries to actually calculate the values in between the original samples. There is plenty of head scratching on how to do this, and once the calculations are made, an "algorithm" (a set of mathematical formulas) is the result. This algorithm is built-into the upsampling chip, and the final set of samples includes new samples generated by application of the algorithm.

Now, of course, Perpetual Technologies' algorithm is proprietary, because not all algorithms are the same. Whatever it is, it is certainly spectacular in its result. My guess is that Fourier analysis has been applied to a number of sounds recorded at very high sampling rate and word length, and then compared to a Fourier analysis of the sounds as they come out after down-conversion to 44.1/16 standard CD signals. Fourier analysis is a way of breaking down a waveform into its various components, called frequency domains. You get number values of the strength of the frequency components in the signal at the same time. The differences between the values in the original recorded signal vs. those in the final 44.1/16 signal represent what is lost in the process. By analyzing different types of music, it seems to me an algorithm could be designed that would put back in what is lost (the differences between the two Fourier analyses). The algorithm might say that if the waveform looks like "A", then "A_1" is added to it, and if "B", then "B_1" is added, and so on. The definition of "A_1" and "B_1" would be based on sitting down and listening to the effects of their addition, with tweaking here and there as to the relative amounts of the frequency domains. So, as a result, the upsampling would not simply smooth out the curve, but actually enhance it with new detail, based on the algorithm's mathematical assumptions. The bottom line is not so much as to whether or not the result is a perfect representation of the original instruments, because no system can do that. It is, rather, does it sound good? Does it give me pleasure? The answer to this is yes on both counts.

To put it another way, the PTs move the sound quality closer to what an actual 96/24 recording would sound like than I have ever heard. It is giving me the opportunity to hear my old CDs in a totally new, and thrilling situation, rather than having to wait until they are all redone as 96/24 discs (from analog tapes or high resolution digital tapes).

The P-1A has speaker correction and room correction as optional features. Depending on your speaker models, you will be able to download the correction algorithm for a fee. The room correction algorithm will be based on having a laptop and microphone sent to you, and following the instructions, you will gather room audio characteristics, and send them back to PT. They write the algorithm, and you install it into the P-1A using the USB port on the P-1A's rear panel. The P-1A has 64 MB of static RAM, so there is plenty of space for complicated mathematics to do their job.

For speaker correction, here is the scenario of how it might work. Just about every speaker on the market has at least two drivers. Two-way speakers have a tweeter and a woofer, three-way speakers have a tweeter, midrange driver, and woofer, and so on. At the crossover points between the drivers, the signal suffers phase shift. This means that, as the signal level declines at crossover, the phase is delayed. It is specified in degrees, such as 10^0, 90^0 (a quarter wavelength), or even 180^0 (a half wavelength). This results in a substantial part of the signal not being time aligned with the rest of the signal, and deteriorates our "you are there" experience. What the P-1A might do is (I don't know for sure, since the actual technique is proprietary), knowing the phase relationships of your particular speakers, slow down the rest of the signal so that it is now in perfect alignment with the phase of the signal that has been slowed down by the crossover network in your speakers. This will provide a very big improvement in the sound. Room correction will adjust the level of various parts of the spectrum that are overemphasized or decreased as a result of the layout of your listening room

The P-1A and P-3A together represent one of the most innovative and important digital audio technologies to come along in several years. They not only have superb sound quality rivaling equipment selling for thousands of dollars more, but have new concepts that no one else offers. The beauty is that PT is just getting started on these products. The algorithms will be improved, and we can download them. They are easily worth the money with no further additions, but the speaker correction and room correction options, coming shortly, make them a must have.

- John E. Johnson, Jr. -

Notes by Paul Knutson

Stacey and John have covered the engineering, so I'll focus on subjective listening impressions, comparisons to other DACs, and finally, suggestions for setting up the P-1A/3A to optimize performance.

P-3A DAC – Listening Impressions

Pardon my statement of the obvious, but the state-of-the-art is advancing exponentially more quickly with digital-to-analog playback than with any other component in the history of home audio reproduction. There's more to come, too, and it's happening fast. A number of respected companies are in the game, but Perpetual Technologies is planning to re-write the rules with a direct-to-consumer business model that offers all the flavor and none of the fat.

The ultimate “winner” of certain pending format wars (SACD vs. DVD-A) remains up-in-the-air, but still, digital audio enthusiasts have much to be happy about right now. Better still, digital-to-analog conversion technology continues to progress while the cost of entry drops like a rock – even for the good stuff.

The Perpetual Technologies P-3A is a perfect example of this phenomenon. Not long ago, you had to consider spending a few thousand dollars to get into digital playback that was acceptably accurate, musical and dynamic, without being colored. Even now, people would have laughed at the suggestion that a mere $699 would get you there . . . but it does.

Used as a stand-alone DAC, the sonic hallmarks of the P-3A are an uncanny sense of realism coupled with astounding clarity, and a dynamic range that can be truly startling. It exhibits that elusive sense of coherence that makes music sound like music, rather than a set of disparate sounds. Reproduction of micro-dynamics, which are really just the small, generally percussive musical details that only emerge when you have a pitch-black sonic background, are also first-rate through the P-3A.

Because of the mind-boggling clarity of the P-3A, the soundstage stretches as wide and deep as the recording permits, with appropriately proportioned images. Listen to Peruvian vocalist Susana Baca’s passionate album *Echo of Shadows* (Luka Bob 72438-48912), and you’ll see that it’s a match made in heaven with the P-3A. Like many of my favorite albums, *Echo of Shadows* was produced by Craig Street. His presence virtually ensures that you’ll be hearing a recording where the vocals, the accompaniment, and the recording venue converge in such a way that it all makes wonderful musical sense. The P-3A is the consummate companion with which to hear these gems.

The Monolithic Sound P^3 Perpetual Power Plant, which Stacey and John discussed earlier, is more than simply an accessory for the P-3A. In my opinion, it’s closer to a necessity if the goal is to extract maximum performance. It isn’t that the P-3A has any true deficiencies using the stock wall-wart power supply, but at a retail price of $699, Perpetual Technologies had to choose their R&D battles with discretion. After extensive listening, I’m certain that the Monolithic Sound P^3 offers the sort of improvement that justifies the added cost. With the additional $349, the total price of the P-3A is bumped to just over a grand, but heck, that’s still comparatively cheap for what you get. From here forward, all my comments about the Perpetual gear presume that the Monolithic is supplying the juice.

I also should mention that, early in my audition, a problem arose that seemingly resided with the P-1A, as it didn’t occur until the P-1A was brought into the system. On two occasions, during a certain musical passage, I heard what sounded like a small electrical “blip”. Immediately following, there was a distinct lack of high frequency detail until I powered down the P-1A/3A and re-set them both. After contacting Perpetual, Jon Lane, the Director of Technical Support, was incredibly responsive and authorized a return, at Perpetual’s expense, to check the units and correct the problem. They were returned to me within a week and have performed flawlessly since. The P-1A and P-3A are right on the edge of audio technology, and as such, one has to expect little problems to pop up at first.

Comparing the P-3A to Other Stand Alone DACs

I spent considerable time comparing the P-3A to a few other acclaimed, high performance

DACs in a variety of price ranges. A stock MSB Link DAC ($399) and Bel Canto DAC-1 ($1,295) were compared to the P-3A in my own system, while the Theta DS ProGenV ($5,600) was the competition in the listening context of a friend's excellent system. These weren't always apples-to-apples comparisons, but they were the best I could do. The Bel Canto, for example, upsamples to 24 bits/96 kHz, while the P-3A performs only 96 kHz upsampling by itself (you need the P-1A to get the 24 bit interpolation). The MSB Link neither interpolates nor upsamples. That said, the comparisons are still valid in the sense that the DACs all sounded subjectively different, which I will report, and with any component, what matters most is how it sounds in your own system.

Here's the signal path of the reference system used for this review, from start to finish:

G&D Transforms Reference One CD transport (Nordost El Dorado power cord)

DH Labs D-75 BNC-to-RCA coaxial digital cable

Monarchy DIP outboard jitter attenuator (Harmonic Technology Pro-AC11 power cord)

Nordost Silver Shadow RCA-to-RCA coaxial digital cable

DAC (typically the TG Audio HSR power cord)

Analysis Plus Silver Oval-In interconnect

Monolithic Sound PA-1 Linestage (HC-2 power supply with TG Audio HSR power cord)

Analysis Plus Silver Oval-In interconnect

Wright Sound Labs WPA 3.5 monoblock power amps

Analysis Plus Silver Oval speaker cable

Silverline Audio SR-15 speakers

Head-to-head, the P-3A easily bettered the stock MSB Link DAC in every parameter. The stock MSB Link is exciting to listen to, a commendable performer and an unquestionably strong value, but also it's a bit rough around the edges. Things that jumped out at me in the comparison were that the P-3A's highs were more present and natural by some margin, while vocals were rendered with truer timbre and wholeness. My listening notes were peppered with the word "coherence" – I was struck by how well the P-3A did it and how the stock MSB Link rather lacked it. This comparison emphasized to me that the P-3A won't find much competition near or below its price point – you must aim higher.

NOTE: I'm referring to the MSB Link in this comparison as "stock" because I've recently had my MSB Link DAC extensively upgraded by Dan Wright, who lives in Portland, Oregon, for a very reasonable cost. Review deadlines prohibit a comparison between the P-3A and the modified MSB Link, but I'll get to that in the future, I promise.

Moving up the price scale, another worthy competitor to the P-3A is the acclaimed $1,295 Bel Canto DAC-1, which features 24/96 upsampling and a slow roll-off 48 kHz filter. Things get interesting here as the DAC-1 is a tremendous performer in its own right (a full review of the DAC-1 will follow in these pages shortly). The Bel Canto DAC-1 is very musical and easy to listen to. It struck me as being somewhat more "beautiful" than the P-3A alone, with a warmer overall tonal character and sweeter highs. The P-3A has the edge in detail retrieval, dynamics and overall realism, all its previously mentioned strong suits. Choosing between these two DACs is like deciding whether to go on a date with Elizabeth Hurley or Tyra Banks – either way you come out a winner. A good litmus test to demonstrate the differences between these two DACs is Charles Lloyd's new (and shockingly good) CD *The Water is Wide* (ECM 1734). Through the Bel Canto, the tone of Lloyd's alto sax on this disc of ballads leans ever so slightly toward soothing and organic. The P-3A brings out more of the brassy, burnished tone, while also more distinctly revealing the multitude of percussive details touched off by the deft Billy Higgins on drums. Whether or not you're a jazz fan, buy this disc and enjoy it often.

In a final comparison, I sought to learn whether the Perpetual could run with one of the "big dogs" of the digital market, the benchmark Theta DS ProGenV. The Theta sells for many times the retail price of the P-3A, and on the surface this comparison seems unfair. Trust me, it's not. We pitted these DACs against each other at a friend's apartment using his reference system, not mine. Tough for an audiophile to admit, but his system is simply better than mine is – it's more resolving and has a more optimal listening space. When the right gear is employed, the system yields a highly involving musical experience.

The Theta vs. the P-3A was a great comparison – these are two wonderful DACs. Each performer took its turn on center stage each earned a standing ovation. As far as differences, from about 80 Hz on up, the Perpetual P-3A simply opened up a clearer window into the musical event and revealed nuance, tonal color, and subtle musical details that the Theta just barely missed. It's that realism thing again – the P-3A is a marvel at extracting the bits. Both DACs have majestic tonal color and dynamic impact. From about 80 Hz and below, however, the Theta reigned supreme in our test, with an authority in the lowest octaves that the P-3A could not quite match. Allow me to dispel any myth that the P-3A is bass-shy, however, because it was only in direct comparison to the Theta that the P-3A was bettered in low bass. Compared to any other DAC in my own system, the P-3A exhibited better performance in the lower octaves. Listen to "She's Already Made Up Her Mind" from *Joshua Judges Ruth* (Curb 10475), Lyle Lovett's folk/gospel album from 1992. That song will put a DAC through its paces – from light cymbal work and plaintive solo vocal at the beginning, through the dynamic drum thwacks about 2/3 of the way through.

When comparing the P-3A as a stand alone DAC to the MSB Link and Bel Canto DAC-1, I didn't initially use the Perpetual P-1A. In the comparison between the P-3A and the Theta DS ProGenV, the P-1A preceded each of these DACs in the signal path. That point segues nicely to my next topic, that being the performance of the P-1A/3A used together.

The P-1A/3A Combo

As a stand alone DAC for normal CD playback, the P-3A soundly established itself in my system as a benchmark for resolution of detail, dynamic impact, coherence and an overall sense of sonic purity. At a price of $699, it'll give the competition fits. Add the P-1A, however, and things get even more interesting.

I'm tired of hearing the term, and I promise not to dwell on it, but the concept of synergy is very real. The P-1A and P-3A are truly *meant* to be used together. Take my word on that. You have a 30-day money back audition period to listen for yourself if you'd like to experience it first-hand. When Perpetual Technologies suggests that the 1A/3A be used in tandem, it's not a veiled attempt to sell more product – it's the stone cold truth.

As arresting as the P-3A is alone, what the P-1A brings to the party is that extra sense of musical refinement, the elusive but palpable sense that what you're listening to just became more real and alive. It enhances your chance to suspend disbelief, even momentarily. The P-1A makes complex music more intelligible by sorting out the mix more distinctly than the P-3A alone. For example, in a great piece of chamber music, you will hear individual virtuosity more intelligibly, which enhances the overall performance. In popular music, solo instruments like a tenor sax or electric guitar bite more sharply when necessary, but whisper more sweetly when asked to as well. You also pick up more of the overtones and undertones, and more of the character and body of the instrument being played. Resolution is further extended than with the P-3A alone, but it's not the kind of resolution that sounds like your car stereo with the bass and treble cranked to 11.

I don't know for certain which aspect of the P-1A's functionality contributes most to what it does – there's the true 24 bit interpolation, the re-clocking of the input signal and subsequent jitter reduction, the speaker and room correction (!) which aren't yet available, but should be soon … the list goes on.

Comparing it to other components that serve similar functions, the P-1A handily outperforms the Monarchy DIP, a fact that was evident when I turned off the resolution enhancement of the P-1A allowing it to function primarily as a jitter reducer. The P-1A also bested the first generation Genesis Digital Lens in my friend's system where the comparison of the P-3A to the Theta was performed. As far as value, $950 may seem like a lot for the P-1A compared to the $699 price of the P-3A, but when you consider all the P-1A does now, and how much more it will do in the future, you quickly realize that it's still a bargain.

The $1,650 P-1A/3A combo, together with the $349 Monolithic Sound P^3 power supply, yield a level of digital playback performance for around $2,000 that was unfathomable only a short while ago. Importantly, the Perpetual products are also fully upgradeable and expandable, which means that as the digital playback express continues to roll along, you will not be left behind. Some of the best and brightest design minds in the industry are with Perpetual, and they are the kind of guys you want on your team. Other companies specializing in leading edge, digital-to-analog conversion are also worthy of your attention – I'll report on some in the near future. As of today, however, the Perpetual Technologies P-1A/3A combo raises the price-to-performance bar in digital playback very high indeed. In fact, forget about the reasonable price for a moment – the bar has simply been raised.

Setting Up the P-1A/3A to Optimize Performance

I mention this in every review, and never has it been more worthy of mention than now: Your system is different than mine, and more importantly, so are your ears and listening preferences. Therefore, please take these suggestions for what they are – merely suggestions and certainly not universal truths.

I compiled a list of setup recommendations during my time with P-1A/3A. During the past few months, I experimented extensively with settings and configurations to learn what worked best in my system. These are worth sharing with you, with the hope that you will go through the same type of experimentation with your own system.

1. Jon Lane of Perpetual Technologies recommended by e-mail that the P-1A be set

to interpolate and output 24-bits, but that the sampling frequency remain at 44.1 kHz. This allows the P-3A to handle the duty of 96 kHz frequency upsampling. After lots of listening, I concur with this recommendation.

2. Each piece comes with two brackets that allow you to install and place the unit vertically on your equipment shelf. This worked fine with the P-1A. However, the P-3A really wanted to be used horizontally in my system. Further, I used three Black Diamond Racing #4 cones under the P-3A for vibration isolation. I didn't use any sort of mass loading on either piece (i.e., putting a brick on top of them), despite that fact that neither weighs a lot, as there was no audible improvement in doing so.

3. Don't use any other type of jitter reducer between your transport and the P-1A – I tried the Monarchy DIP and it robbed the music of life. The Monarchy DIP is a good product on its own when the P-1A is not available, but in tandem with the P-1A, it does more harm than good.

4. This is a big one . . . I do not recommend using the factory-supplied I^2S cable between the P-1A and P-3A. Sure, I realize that theoretically this should be an ideal interface, but that wasn't what I heard. The cool thing is that you can connect both the I^2S cable and a coaxial digital 75 Ohm cable between the P-1A and P-3A, and then toggle between the two to perform your own evaluation. Heck, throw in an AES/EBU balanced cable if you have one, too. Speaking with a friend on the topic, I think it comes down to the fact that the factory-supplied cable is the culprit and not the I^2S technology. I didn't have any other I^2S cables with which to compare the one supplied by Perpetual, but rumor is that Perpetual is working with a highly-regarded wire company to come up with a better I^2S cable. In lieu of the I^2S connection, I had great results connecting the 1A and 3A with the inexpensive Nordost Optrix coaxial digital cable, and even better was the awesome Nordost Silver Shadow.

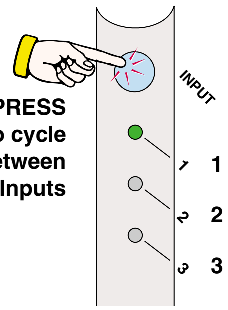
5. Give the P-3A a ton of time to break-in – in the realm of 200 hours. Really, that's how long it took in my system before the audible improvement stopped. The P-1A took less time, along the lines of 100 hours, until it sounded its best.

6. Use power conditioning, especially with your digital front-end, if you don't already. The Perpetual gear loves it. I used units from Sound Application, ExactPower, and PS Audio. This isn't going to become a power conditioner review, but suffice to say that they all help bring out the best that the P1A/3A have to offer.

- Paul Knutson -

P•1A Mode Selection Routines *(Updated 7/01 to include routines for new code. These instructions supercede owner's manual.)*

STEP A — Select INPUT

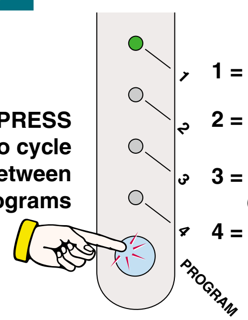


PRESS to cycle between Inputs

1 = I^2S
2 = AES/EBU
3 = Coaxial S/PDIF

Green = Active/Locked
Red = Inactive/Unlocked

STEP B — Select PROGRAM



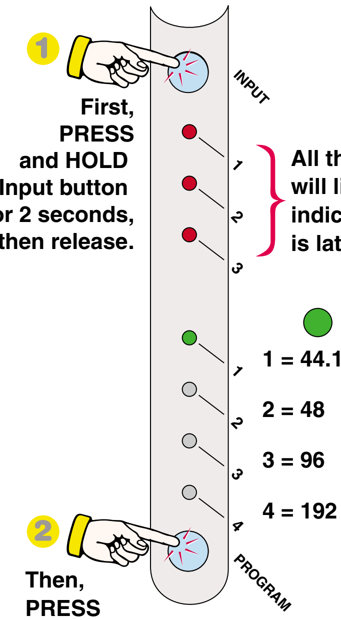
PRESS to cycle between Programs

1 = Bypass
2 = CD Res Enhance
3 = Correction *(if available)*
4 = Both 2 + 3 *(if available)*

Operation with CD Resolution Enhancement
When no speaker correction software is installed, pressing the Program button toggles between Bypass (LED 1) and CD Resolution Enhancement (LED 2) modes.

Operation with Speaker Correction
Once the P•1A has been upgraded to include speaker or speaker/room correction, pressing the Program button cycles the P•1A output between all four modes: Bypass (LED 1), Resolution Enhancement (LED 2), Correction only (LED 3), and Both Correction and Resolution Enhancement (LED 4).

STEP C — Select SAMPLE RATE (SRC)



1 **First, PRESS and HOLD Input button for 2 seconds, then release.**

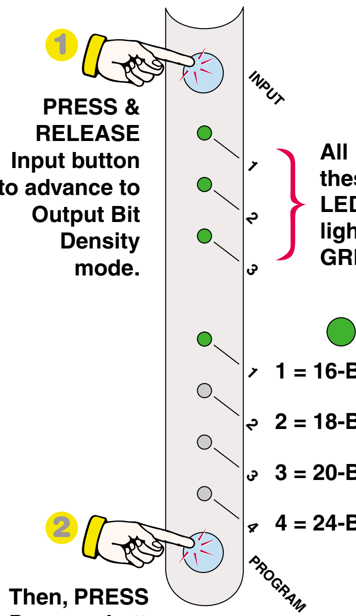
All three LEDs will light RED indicating mode is latched

1 = 44.1
2 = 48
3 = 96
4 = 192 *(if available)*

2 **Then, PRESS Program button REPEATEDLY to cycle through Sample Rates**
- **Green = selection**

Note:
- *Each of the 3 Inputs has its own Sample Rate setting.*

STEP D — Select OUTPUT BIT DENSITY



1 **PRESS & RELEASE Input button to advance to Output Bit Density mode.**

All these LEDs light GREEN

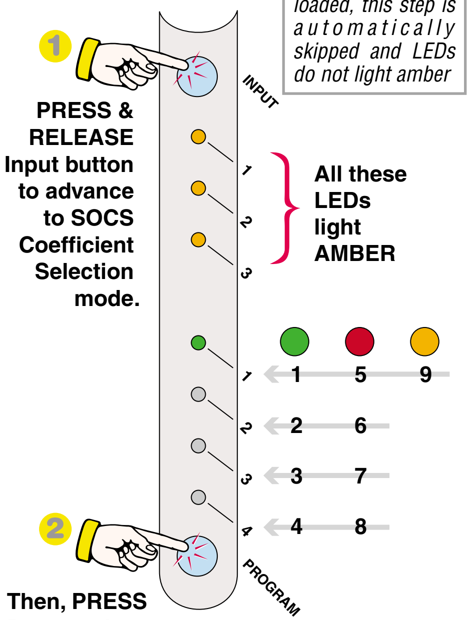
1 = 16-Bit
2 = 18-Bit
3 = 20-Bit
4 = 24-Bit

2 **Then, PRESS Program button REPEATEDLY to cycle through Bit Density mode to desired density**
- **Green = selection**

Note:
- *All Input modes use the same Output Bit Density setting.*

STEP E — Select SOCS COEFFICIENT

If there is no SOCS loaded, this step is automatically skipped and LEDs do not light amber



1 **PRESS & RELEASE Input button to advance to SOCS Coefficient Selection mode.**

All these LEDs light AMBER

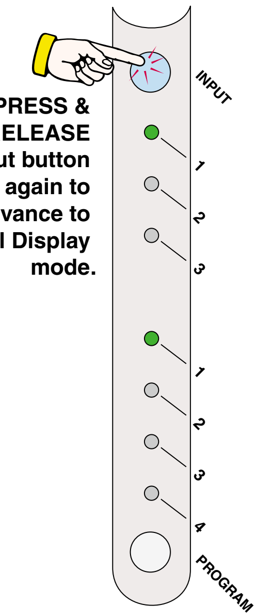
1	**5**	**9**
2	**6**	
3	**7**	
4	**8**	

2 **Then, PRESS Program button REPEATEDLY to cycle through SOCS Coefficient modes**
- **Green = Coefficients #1–#4**
- **Red = Coefficients #5–#8**
- **Amber = Coefficient #9**

Notes:
- *Once you reach #9, the cycle continues with #1, #2, #3, etc.*
- *All Input modes use the same SOCS Coefficient setting.*

Important feature:
- *As you select each SOCS Coefficient, you can immediately hear the sonic results without having to leave this step. This feature makes it easy to do an A-B comparison between two coefficients.*

STEP F — Return to Normal Display



PRESS & RELEASE Input button once again to advance to Normal Display mode.

When you press the Input button the display will return to Normal with green LEDs indicating the active Input and Program selections.

SRC, Output Bit Density and SOCS Coefficient selections are all stored in memory.

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